



Anisotropy and sound propagation in glass wool

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ability (MAP) theory to produce a background estimate. The data model is guided by the measurement physics and the scene model is based on a Markov random field assumption. Both models are augmented to accommodate anomalies. In this seminar the MAP estimator is described, along

with practical issues associated with algorithm initialization, simulated performance with both white and colored-noise inputs, and real data examples. [Work supported by SPAWAR, under Air Force Contract F19628-95-C-0002.]

Contributed Posters

These posters will be on display in the Poster Gallery from Thursday to Friday, 18–19 March. Authors will be at their posters from 2:00 p.m. to 4:00 p.m. on Friday, 19 March.

5aUWb11. Identification of the radiating sources in underwater acoustics. Bourennane Salah and Bendjama Ammar (URA CNRS 2053, Quartier Grossetti 20250, Corte, France)

In the context of the narrow-band or wideband array processing problem, in this paper a robust algorithm is developed to improve the accuracy of the estimation of the direction of arrival of the sources. It is well known that when the noise cross-spectral matrix is unknown, these estimates may be grossly inaccurate. The band noise spectral matrix, the classical noise model, is used. By means of a linear operator and an iterative algorithm for estimating the spatial correlation length, a new estimator for the direction of arrival of the sources is developed. The proposed algorithm is based upon a particular partition of the received signal vector which leads one to obtain an approximation of the noise subspace matrix without eigendecomposition. Then using both a propagation operator and non-eigenvector algorithm to estimate the projection matrix, a new robust algorithm is developed for the source characterization problem in the presence of noise with an unknown cross-spectral matrix. It will be shown that the performance of bearing estimation algorithms improves substantially when this robust algorithm is used. Experimental data results are presented for the band noise spectral matrix.

5aUWb12. Three-dimensional auditory display of passive sonar data. Suzanne Richardson (Schlumberger, 8311 North FM 620 Rd., Austin, TX 78726, srichard@austin.apc.slb.com) and Charles Loeffler (Univ. of Texas, Austin, TX 78758)

Submarine sonar operators typically use visual and monaural audio outputs to locate obstacles, ships, and animals in the surrounding ocean. Recently, the effectiveness of using 3-D, rather than monaural, sonar audio outputs was investigated. A multi-channel signal processing system has been developed that converts sonar signals collected by elements of a hydrophone array into binaural signals for the left and right ears, which provide accurate 3-D aural imaging of an underwater acoustical environment. An operator who listens to the binaural signals via stereo headphones will perceive that he is immersed in the underwater acoustical environment. The system incorporates two banks of FIR filters (one for the left ear and one for the right ear) to process the sonar data. The FIR filters simultaneously beamform the hydrophone data and filter the data with head-related transfer functions in order to create binaural signals that provide 3-D sound. [Work supported by Applied Research Laboratories: U. T. Internal Research and Development.]

FRIDAY AFTERNOON, 19 MARCH 1999

POSTER GALLERY, 12:00 TO 6:20 P.M.

Posters from various technical sessions remain on display in the Poster Gallery.

Posters from sessions which contain both lecture and poster presentations will be attended by the authors as listed below.

2:00–4:00

5aNSa8	Bruyninckx, Willy	Measurement methodology for sound power levels of industrial noise sources
5aPac11	Rath, M.	Inline process monitoring of thermosets by ultrasonic measurements in a compression mould
5aPac12	Flannery, C. M.	Accurate noncontact measurement of elastic properties of ceramics using acoustic microscopy and laser ultrasonics
5aPac13	Kalinchuk, Valery V.	Some dynamic properties of non-homogeneous thermoelastic pre-stressed media
5aPac14	Kalinchuk, Valery V.	On the investigation of the stressed state of the elastic and piezoelectric bodies
5aUWa13	Furduiev, Alexandr	Internal wave imprint in the oceanic ambient noise
5aUWa14	Furduiev, A. V.	Ocean monitoring by using underwater sound channel as a feedback loop
5aUWb11	Salah, Bourennane	Identification of the radiating sources in underwater acoustics
5aUWb12	Richardson, Suzanne	3-D auditory display of passive sonar data

Also, the following poster sessions are scheduled:

Poster Session 5pPPb

Poster Session 5pSCa

Poster Session 5pSCb

Session 5pAA

Architectural Acoustics: Building Acoustics III: Acoustical Measurements

K. Heinrich Kuttruff, Cochair

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Angelo Campanella, Cochair

Campanella Associates, 3201 Ridgewood Drive, Columbus, Ohio 43026, USA

Chair's Introduction—1:55

Contributed Papers

2:00

5pAA1. A system for measuring the directional parameters of room acoustics. Hiroyuki Okubo, Masamichi Otani, Ryo Ikezawa, and Setsu Komiyama (Japan Broadcasting Corp. (NHK), 1-10-11 Kinuta, Setagaya-ku, Tokyo, 157-8510 Japan)

A new system for measuring the directional parameters of room acoustics has been developed. The system consists of five omni-directional microphones, audio interfaces, DSPs (digital signal processors), and control software. This software synthesizes the directional pattern of the microphones, and calculates the directional parameters automatically. The lateral component (LC), front-to-back ratio (FBR), and left-to-right ratio (LRR) of directional room impulse responses are easy to measure with this system. The system was used to take measurements in two halls. The first hall was a concert hall, with the measurements of impulse responses taken at 72 points. The second was a multipurpose hall, the shape of which could be altered three ways (for concerts, conventions, and exhibitions), and the measurements were taken at 12 points for each shape. The total of 108 data points from the four room shapes represents the range of room types. The directional parameters are sensitive to room shape, wall materials, and the positions of seats. These data may also be useful for studies on the correspondence between the objective description and subjective evaluation of the acoustical quality of rooms.

2:20

5pAA2. Frequency response measurement with composed audio test signal. Mladen Maletic, Hrvoje Domitrovic, and Ivan Djurek (Faculty of Elec. Eng. and Computing, Dept. of Electroacoustics, Unska 3, HR-10000 Zagreb, Croatia, mladen.maletic@fer.hr)

The composed audio test signal (CATS) is formed by mixing three mutually independent composed saw signals (CSSs). The CSS sequences are a special type of pseudorandom signal. The CATS closely follows the characteristics of natural signals, and its behavior is controlled by the well-defined and configurable parameters. The amplitude probability density function of the CATS follows Gaussian distribution. Furthermore, second and third CSS sequences are generated using the perturbed parameters of the first one. Pseudorandom noise can have a spectrum and amplitude distribution similar to that of white noise. The CATS methods employ efficient cross correlation between input and output to recover the periodic impulse response (PIR) of the system being measured. The PIR can be considered as the impulse response only if the CATSs period equals or exceeds the duration of the system impulse response. The frequency response can then be found by taking the discrete Fourier transform of the impulse response. The CATS can be used as a test signal with two-channel analyzers. The averaged cross spectrum is divided by the averaged autospectrum of the input channel to obtain the frequency response.

2:40

5pAA3. Fluctuation of room acoustical parameters on small spatial intervals. Diemer de Vries and Jan Baan (Lab. of Acoust. Imaging and Sound Control, Delft Univ. of Technol., Delft, The Netherlands)

The Delft Acoustics group has introduced array technology in room acoustic measurement practice: instead of at a limited number of "representative" places, impulse responses are measured along an array of closely sampled microphone positions. By displaying the responses as a visual entity, the wave character of the sound field is clearly revealed. The data form a base for further processing, enabling the separation and analyzing of the different wave-field components. The sound fields in several concert halls have been measured this way. From impulse responses recorded with 0.05-m spatial separation, the common room acoustical parameters have been determined. It appears that, even between the ten positions in front of one and the same seat, unexpected fluctuations occur, e.g., when measured in octave bands, the clarity index varies over 1 dB on this interval, the lateral energy fraction 0.2 (i.e., between 0.1 and 0.3). For the broadband versions, these variations are approximately halved, still being significant. These results explain why the predictive value of these parameters for the perceptual quality of the acoustics on a (group of) seat(s) is limited. Instead of at one point, the values at several adjacent points should be considered.

3:00

5pAA4. Influence of time variance on room impulse response measurement. Fumiaki Satoh (Dept. of Architecture, Chiba Inst. of Technol., Tsudanuma 2-17-1, Narashino-shi, Chiba, 275-0016 Japan), Yoshito Hidaka (Tohwa Univ., Chikushigaoka 1-1-1, Minami-ku, Fukuoka-shi, Fukuoka, 815-8510 Japan), and Hideki Tachibana (Tokyo Univ., Roppongi 7-22-1, Minato-ku, Tokyo, 106-8558 Japan)

The measurement of room impulse response is made under the assumption that the sound propagation system is time invariant. Actually, however, the air in a room is usually moving and the temperature is changing to a greater or lesser extent. In order to examine the influence of such time variance on the measurement of room impulse response, experimental investigations were performed in a reverberation room in which the air was excited by fans and in a concert hall in which the air-conditioning system was operated. Impulse response measurements were performed by the maximum-length sequence (MLS) method and the sweep pulse (SP) method and these results were compared. From the results, it has been found that the reverberation decay tends to become steeper by repeating the impulse response measurement to get high signal-to-noise ratio and the SP method is more robust than the MLS method against the influence of time variance of the atmospheric condition in a room.

5pAA5. Time variances with MLS technique: Theoretical considerations concerning rotating microphones in building acoustics. Wieland Weise and Alfred Schmitz (Appl. Acoust., Physikalisch-Technische Bundesanstalt, Bundesallee 100, 38116 Braunschweig, Germany)

For some time now, the technique of maximum length sequence (MLS) measurement has been successfully applied in building acoustics. Rotating microphones which are otherwise useful, especially for the determination of sound reduction, cannot be used here because the time invariance of the system would be violated. The measurement errors arising when rotating microphones are used in experiments were already investigated previously. However, in view of the future process of standardization of measuring techniques in building acoustics, theoretical understanding is necessary. Theoretical analyses are presented which exhibit the deviations of the measurements. Emphasis is placed on the differences arising when a synchronous or asynchronous relation between microphone round trip time and MLS period is chosen.

3:40–4:00 Break

4:00

5pAA6. Measurements of anechoic room characteristics. Miroslava A. Milosevic and Dejan G. Ciric (Faculty of Electron. Eng., Univ. of Nis, Nis, Yugoslavia)

A simple measurement system with a computer as the main unit is developed for the measurements of the characteristics of an anechoic room, such as measurements of the sound-pressure decay in space, the sound-pressure decay in time, and the sound insulation. The measurements are performed in two steps. The room response signals are first recorded and stored in the PC memory. Then, these signals are processed by designed software depending on the type of measurement. The departure of sound-pressure decay in space from the ideal characteristic is observed.

The sound-pressure decay in time is obtained by the integrated-impulse method using the maximum-length sequence technique. The sound insulation is measured only for a specially designed door between the anechoic and the control room because of its insulation, which is rather low in regard to the other parts of an anechoic room. The measurements performed in the anechoic room of the Faculty of Electronic Engineering in Nis have shown that the departure of sound-pressure decay in space from the ideal characteristic is rather below the limit of 1 dB in frequency range from 50 Hz upward. The door sound insulation is above 55 dB.

4:20

5pAA7. An Internet-oriented system for acoustic measurements of sound fields. Masatugu Sakurai, Shinichi Aizawa (Yoshimasa Electronic, Inc., 1-58-10, Yoyogi, Shibuya, Tokyo, 151-0053 Japan, info@ymec.co.jp), and Yoichi Ando (Kobe Univ., Nada, Kobe, 657-8501 Japan)

A new diagnostic system was developed, based on the model of the auditory-brain system. The system, including the AD and DA converters, works on the PC for Windows, thus there is no need for special additional devices. Further, the system is available on the Internet, where a user can obtain further information about the system, and utilize it easily in practice by downloading a demonstration program. After obtaining the binaural impulse responses, four orthogonal factors, including the SPL, the initial time delay gap between the direct sound and the first reflection, the subsequent reverberation time, and the IACC are analyzed. Then, these factors may be used for the calculation of the scale values of both global and individual subjective preference. In addition, two more factors, τ_{IACC} and W_{IACC} , in the interaural cross-correlation function are figured out for evaluating the image shift or balance of sound fields and the apparent source width (ASW), respectively. Also, the effective duration of ACF, τ_e , and fine structures of autocorrelation function of sound signals, including the value of first maximum, ϕ_1 , and its delay time, τ_1 , are analyzed. This system may be utilized for automatic control of sound fields by electroacoustic systems also.

Contributed Posters

These posters will be on display from Monday to Wednesday, 15–17 March. Authors will be at their posters from 10:00 a.m. to 12:00 noon on Wednesday, 17 March.

5pAA8. Four-microphone-array measurements combined with geometrical room acoustic simulation technique. Jörg Becker, Markus Sapp (Inst. of Commun. Eng., Aachen Univ. of Technologie, 52056 Aachen, Germany), and Oliver Schmitz (Aachen Univ. of Technologie, 52056 Aachen, Germany)

For a revision of the acoustics of enclosures it is important to analyze the existing acoustical mechanisms. Therefore, measurement of room impulse response functions is a usual standard. The temporal progression of the incident sound can be determined from these impulse responses. Single number quantities can be evaluated, too. However, the knowledge of the direction of the first reflection is also of prime importance. Computer simulation of the room acoustic is a powerful tool to evaluate these directions but it is only reasonable if the geometry of the room is well known. Often it is difficult to get detailed information of the geometry, especially if there are curved walls. This paper focuses on the evaluation of spatial information of sound fields with a four-microphone-array measurement system and the post-processing with algorithms taken from geometrical room acoustic simulation. With multidimensional sound field analysis it is possible to get the direction of the important reflections. Virtual sound sources can be calculated. With a back tracing algorithm the history of higher-order reflections can be determined or the geometry of

the room model and the results of computer simulation can be verified by a comparison of measured and calculated VSS.

5pAA9. On the accuracy of the assessment of room acoustical parameters using MLS technique and ray-tracing simulation. Marcio Henrique de Avelar Gomes and Samir Nagi Yousri Gerges (Universidade Federal de Santa Catarina-Depto. de Eng. Mecanica-Laboratorio de Vibracoes e Acustica-cx. postal 476-Florianopolis, SC 88040 900, Brazil, avelar@gva.ufsc.br)

Advances in psychoacoustics through the years have made it possible to evaluate the acoustical quality of a room, based on several numerical parameters that have been developed. Even though there is not total agreement about which parameters are truly important, some of them are accepted by most of the acousticians. Most of these parameters can be determined from the impulse responses of a room. This work presents a comparison between results simulated using a commercial ray-tracing computer program, and measured with one of the most modern techniques of measuring impulse responses, which uses a maximum length sequence (MLS) as the driving signal. Limitations concerning the use of the measuring technique, and the ray-tracing program, as well, are analyzed and discussed.

Session 5pBB

Biomedical Ultrasound/Bioresponse to Vibration: Low Intensity/Low Frequency Ultrasound

E. Carr Everbach, Cochair

Engineering Department, Swarthmore College, 500 College Avenue, Swarthmore, Pennsylvania 19081-1397, USA

Gail ter Haar, Cochair

Physics Department, Royal Marsden Hospital, Sutton, Surrey SM2 5PT, UK

Chair's Introduction—2:15

Invited Papers

2:20

5pBB1. Therapeutic ultrasound—An historical overview. Gail ter Haar (Phys. Dept., Royal Marsden Hospital, Sutton, Surrey SM2 5PT, UK)

Ultrasound was used with therapeutic intent long before its potential as an imaging modality was contemplated. Initially it was thought of as acting solely through its heating ability, but more recently there has been scientifically convincing evidence that beneficial effects can be induced in the absence of biologically relevant bulk temperature rise, thus indicating that nonthermal mechanisms may also play a part in ultrasound treatments. Apart from the high-intensity, surgical uses (operating at peak intensities around 2 kW cm^{-2}), ultrasound has been applied amongst other things to promote wound healing, to stimulate bone repair, to accelerate healing in tendon injuries, and as a thrombolytic agent. The majority of therapeutic ultrasound techniques operate at intensities of 500 mW cm^{-2} or less (spatially averaged intensities). It is interesting to compare the different techniques and modes of delivery in an attempt to understand the mechanisms of action that produce beneficial biological effects.

2:40

5pBB2. Enhancement of fibrinolysis by low-intensity c.w. ultrasound. Charles W. Francis, Valentina Suchkova (Vascular Medicine Unit, Univ. of Rochester Med. Ctr., Rochester, NY 14642), and E. Carr Everbach (Swarthmore College, Swarthmore, PA 19081)

Continuous-wave ultrasound at intensities below 4 W/cm^2 (SATA) and frequencies around 1 MHz has been shown to accelerate enzymatic fibrinolysis *in vitro* and in some animal models. The mechanism for this ultrasound accelerated thrombolysis (UAT) is unknown, although heating is unlikely. Hyperbaric studies have shown that approximately half of the acceleration is removed with overpressures of 3 atm or more, implicating cavitation mechanisms as dominant contributors to UAT, if not the sole ones. Detection of bubbles and cavitation activity in UAT via passive and active detectors corroborates this view. Recent studies using 40-kHz cw ultrasound at intensities as low as 0.25 W/cm^2 show promise for eventual clinical applications, including insonification through bone without significant heating. [Work supported by NIH.]

Contributed Papers

3:00

5pBB3. Ultrasonic-induced phase transitions of micrometer-size droplets. Oliver D. Kripfgans, J. Brian Fowlkes, and Paul L. Carson (Univ. of Michigan, Dept. of Radiol., Kresge III R3315, Ann Arbor, MI 48109-0553, oliver.kripfgans@umich.edu)

Droplets for various applications were generated by mixing albumin (bovine) with saline and a low boiling point liquid (dodecafluoropentane). The resulting emulsion contains small droplets with a lower size range on the order of a few micrometers. The type of mixing process was found to determine the range of droplet sizes. The albumin, as a surface active agent, is assumed to build a shell around the droplets and prevents coalescence, since mixtures without albumin did not produce stable droplets. Increases in temperature of the resting host fluid showed that droplets could be superheated well above their natural boiling point. It is assumed that this property is caused by an increase in pressure within the droplet due to surface tension. The superheated state could be overcome by the application of ultrasonic pulses and/or hydrodynamical action (flow). The presence of ultrasound contrast agents allowed ultrasonic-induced phase transitions at the natural boiling point. Droplets producing bubbles for transpulmonary applications have been studied previously. The interest here was primarily in larger bubbles for therapeutic applications and to

serve as point beacons for aberration correction. [Work supported in part by PHS Grant No. R01 HL54201 from the National Heart, Lung and Blood Institute.]

3:20

5pBB4. Ultrasonic technique for the investigation of structural properties of biological fluids. Algirdas Voleišis, Rymantas J. Kažys, Liudas Mažeika, Reimondas Šlitteris, and Birutė Voleišienė (Ultrasound Res. Ctr., Kaunas Univ. of Technol., Studentų str. 50, LT-3031, Kaunas, Lithuania, ulab@tcf.ktu.lt)

Biological fluids are specific objects for acoustical investigation due to the wide spectra of relaxation processes, especially nonstationary fluids such as blood during its coagulation process. The proposed method combines measurement of ultrasound attenuation over a frequency range and, ultrasound velocity dispersion being negligible, precise measurement of velocity variations at fixed frequency. In the dynamic spectroscopy method the wideband ultrasonic signal transmitted through the media is digitized with a sample rate of 200 MHz, averaged, and processed by a PC. Ultrasound absorption frequency dependency with intervals of 1 min is determined from amplitude spectra. Using small volume (1 ml) cell with

multiple reflections, in the range of 2–17-MHz, diffraction corrections and ultrasonic attenuation were determined in low-absorptive standard liquid, conservative, and native coagulating blood. The clot formation process in the native blood is also monitored at frequencies 5 or 10 MHz using the time-of-flight method with the acquisition of the results. In the cell of 0.2 ml volume sensitivity of 0.2 ns is achieved. Cell parameters and influence of the pulse bandwidth are analyzed. According to the ultrasound velocity variations, four typical blood coagulation phases were observed during 30 min, less expressed being the first phase (0–2 min).

3:40

5pBB5. Are there gas cavities in red blood cells? Dmitry A. Selivanovsky, Igor N. Didenkulov, and Olga V. Kostina (Inst. of Appl. Phys., 46 Ulyanov St., Nizhny Novgorod, 603600, Russia)

Many of marine single-cell organisms (phytoplankton) have gas cavities which allow them to keep their life cycle. Existence of gas cavities can essentially have an influence on acoustic and mechanical properties of cells. It seems reasonable to suppose that such cavities can also exist in erythrocytes, life cycle of which is related to gas exchange. The present work was aimed on measurements of acoustomechanical properties of human red blood cells to reveal their gas cavities. The velocities of erythrocyte sedimentation (VES) in natural conditions and under compression/decompression were measured. The VES was found to increase considerably under or after compression/decompression. The effect was observed both for blood from healthy and sick people. Special microscope observations of sedimentation process allowed findings that compression/decompression changes the natural orientation of erythrocytes: after compression/decompression their main planes become vertical instead of horizontal for natural conditions. Such a phenomenon can be explained with the model of the red blood cell as a body with gas cavity. A compression/decompression changes the balance between centers of gravity and buoyancy making cells rotate and consequently to decrease the viscous force. [Work supported in part by RFBR, Russia.]

4:00

5pBB6. The acoustic system for investigation of the temporomandibular joint pathology. Waldemar Lis, Roman Salamon, and Jozef Zienkiewicz (Dept. of Acoust., Tech. Univ. of Gdansk, 80-952 ul. G. Narutowicza, 11/12 Poland, wall@eti.pg.gda.pl)

The temporomandibular joint is the most complicated joint from the anatomic standpoint. Diagnosing it is still a problem to be solved. Modern top quality devices do not provide a complete solution—the barrier here is the high price of a single examination. For this reason, new, cheap methods of diagnosing the temporomandibular joint are being sought. One of these methods is diagnosing on the basis of passive listening for acoustic signals emitted by the joint in motion. This paper presents the application of the FFT and wavelet method for the purpose of examining the acoustic signal analysis of the temporomandibular joint. The advantages and disadvantages of both methods, the Fourier method and wavelet method, are discussed. Finally, the results obtained by the Fourier analysis in time window and by a wavelet method are presented and compared.

4:20

5pBB7. The velocity of ultrasound propagation through brain tissue at low ultrasound frequencies. Igor Zoric (Faculty of Mining and Petroleum Eng., Pirotijeva 6, HR 10000 Zagreb, Croatia, izoric@rudar.rgn.hr), Bojan Ivancevic, and Kristian Jambrosic (Faculty of Electrotechnic Eng. and Computing, HR 10000 Zagreb, Croatia)

The acoustic properties of the brain tissue have been in the focus of much scientific research. These properties determine the possibilities of the application and development of the ultrasound methods, as well as the development of the equipment used by the neurosurgeon during the resection of the brain tissue. The results of the measurements of these acoustic properties done at megahertz frequency range have been published. Although the velocity of ultrasound wave propagation does not directly influence the distribution of the ultrasound field within the cranium, it enables the determination of the specific resistance, which has an important role in the distribution of the ultrasound field within the cranium. Using the published results and the conditions of the measurements that have been done, the velocity of ultrasound propagation for the frequency range at which cavitation ultrasonic surgical aspirators work has been estimated. After that the measurement of the velocity of the propagation has been done at the frequency of 24 kHz. The values of both the estimated and the measured velocities of the ultrasound propagation prove the validity of the assumptions on which the estimation is based on.

4:40

5pBB8. Modification of chamber resonance through active mass control. James S. Martin and Peter H. Rogers (Georgia Inst. of Technol., School of Mech. Eng., Atlanta, GA 30332-0405)

A traveling wave chamber was constructed for the controlled insonification of rodents in order to test the physiological effects of intense underwater low-frequency sound. In initial testing, this chamber exhibited a sharp resonance in the band of interest. This resonance was found to be dictated by the mass of the piston driver and the stiffness of the chamber's contents. The presence of the resonance significantly reduced the operational bandwidth of the chamber. Conventional approaches to shifting the resonance out of the band of interest by directly altering piston mass and/or armature stiffness were found to be impossible. An open-loop active control system was devised which utilized piston-mounted accelerometers and summed their output with the drive signal prior to a power amplifier. Since the driver is well characterized as a force transducer, the result of this control scheme is that the piston mimics a significantly lighter structure with no reduction in the actual piston mass. The operational bandwidth of the chamber was increased with a minimum of additional hardware and negligible additional power consumption. Loop gain can be limited by stability limits imposed by the phase response of the power amplifier. This scheme may also have applications in transducer design.

Session 5pEA

Engineering Acoustics: General

P. K. Raju, Cochair

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Marinus M. Boone, Cochair

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Chair's Introduction—1:55

Contributed Papers

2:00

5pEA1. Using tubephones for wideband auditory stimulation in brain research. Matti Airas, Antti Jarvinen (Lab. of Acoust. and Audio Signal Processing, Helsinki Univ. of Technol., P.O. Box 3000, FIN-02015 HUT, Finland), and Esa Piirila (Sample Rate Systems Oy, Tampere, Finland)

This study was motivated by the need for a high-quality auditory stimulation system for brain research purposes. Magnetic imaging techniques such as functional magnetic resonance imaging (fMRI) and magnetoencephalography (MEG) prevent the use of typical headphones. In fMRI, a high magnetic field does not allow placing magnetic materials inside the imaging system. In MEG, the extremely weak magnetic fields present in the human brain are measured and any magnetic material in the magnetically shielded room would corrupt the results. In these cases, one possible solution is to place electromagnetic transducers outside of the measurement room. Sound has then to be transmitted to the patient's ears by tubes which produce a highly distorted response. It is shown that by combining acoustical design with analog and digital equalization, wideband flat frequency response is achieved. The design and construction of the apparatus are described and problems like the effect of nonlinear distortion of the transducers are discussed. Practical applications of the system are shown and other possible constructions are compared with the implemented one.

2:20

5pEA2. A parametrized model of psychoacoustical roughness for objective vehicle noise quality evaluation. Robert Hoeldrich and Martin Pflueger (Joanneum Res., Schiestattgasse 14/b, A-8010 Graz, Austria)

In the assessment of vehicle interior noise quality, sound characteristics caused by modulation play an important role because they contribute significantly to the perceived annoyance. The proposed parameterized model for the sensation of psychoacoustical roughness was developed as part of a research program with the aim of establishing an on-board analysis system for vehicle interior noise quality based on objective sound parameters. The roughness model can be adjusted by model parameters to calculate versions of roughness, thereby accentuating different psychoacoustical assumptions. The model is based on the excitation-time pattern of the interior car sound which is recorded using an artificial head system. From the excitation-time pattern, a set of specific roughness parameters in overlapping critical bands is obtained utilizing different weighting functions for the envelope spectrum in each band and various correlation methods between adjacent bands. The specific roughness parameters in critical bands are superimposed in several ways, resulting in a set of modulation parameters. The model was successfully tested not only in predicting roughness assessments, as reported in a companion paper, but it also proved to result in one valid objective parameter for the description of vehicle interior noise quality.

2:40

5pEA3. Psychoacoustic measurement of roughness of vehicle interior noise. Martin Pflueger, Robert Hoeldrich (Joanneum Res., Schiestattgasse 14/b, A-8010 Graz, Austria), Franz Brandl, and Werner Biermayer (AVL List GmbH, A-8020 Graz, Austria)

In analyzing vehicle interior noise there exists hardly any correlation between the subjective impressions of test persons and available roughness parameters. It has become necessary to develop a new roughness model for these sounds and to set up a subjective database as a basis for the analytical description of interior noise quality. This paper focuses on psychoacoustical experiments for the assessment of roughness by using different methods and different kinds of vehicle interior sounds. The analysis of the results shows that small experimental manipulations lead to a large variety of different results. The different meanings of the word roughness have to be taken into account. Each person has his or her own interpretation of roughness differing between the phenomena of roughness, *r*-roughness, rumble, fluctuation strength, etc. Another important point is the reduction of accidental influences of psychoacoustical parameters, like, for example, the influence of loudness on the perception of roughness. By using a special headphone-subwoofer playback system in the laboratory, it is possible to achieve ratings comparable with field experiments. As reported in a companion paper, a new generalized psychoacoustical model of modulation parameters was successfully tested with the obtained roughness rankings.

3:00

5pEA4. RPG diffusers used for noise screening applications. Franco G. M. Bertellino, Luca Stantero, and Antonio Massacesi (Microbel s.r.l., via Livorno, 60, Torino, Italy, microbel@envipark.com)

The paper deals with the performances of noise barriers designed using the reflecting phase grating diffuser theory, which is based on the interference produced by slots of different depths. Usually such structures are used to improve sound diffusion in recording rooms, but they can be suitable also to reduce highway and railway noise for their capability of scattering sounds. A theoretical approach has been used to project their shape using a self-developed VB application, able to predict the scattered sound distribution as well. The theoretical results have been verified with a physical scale model in an anechoic room: insertion losses in several configurations have been measured, together with intensity measurements of the scattered acoustic field. Experimental results seem to match quite well those obtained using the simulation program.

5pEA5. Multi-purpose apparatus for evaluation of noise transmitted through air ducts. Antonino Di Bella and Roberto Zecchin (Dipartimento di Fisica Tecnica, Univ. of Padova, Via Venezia 1, I-35131 Padova, Italy, labacus@iftpd1.iftpd.unipd.it)

The noise generated in HVAC systems by fans or air turbulence is propagated along ducts and, through duct walls, can be irradiated in rooms; moreover duct lines can easily transmit the sound produced in noisy environments from one room to another, vanishing the sound insulation of walls which they are passing through. The knowledge of sound insulating properties of duct walls is fundamental for a good system design, including unitary attenuators and active noise control devices. In this work a multi-purpose apparatus for evaluation of noise transmitted through air ducts in a reverberant room is presented. This apparatus allows one to measure, with the same equipment, the sound power level emitted from a short length ducts specimen in a reverberant room, as well as linear sound attenuation and break-in and break-out transmission loss. A further application of the same apparatus is the measurement of sound insulation and normal incidence impedance of small plane elements of isotropic or composite materials.

5pEA6. Annoyance of the induction machine noise radiation. Cüneyt Öztürk and Ayдын Bahadır (Türk Elektrik Endüstrisi AP, R&D Dept. Davutpapa, Litros Yolu, Topkapı, 34020, Istanbul, Turkey, cozturk@tee.com.tr)

Designing and developing products to enhance the favorable and diminish the undesirable features of their sounds are the prime concerns of noise control engineers. Development of new products that might be based on the understanding of the physics of sound, how sound is perceived by the listeners, and how special features of sound determine our reactions are all significant issues of noise engineering. Noise problems of the induction machines mostly arise during the early stages of design and development. The main energy source in an electric motor is the magnetic field. Maxwell stresses act on the inner stator surface bore, giving rise to spectral distribution over the stator as a function of time. These forces excite the stator's lamination and housing, which have distributed mass and damping. This study describes the noise radiation characteristics of the induction machine with special emphasis on the annoyance issue that varies behind the declared emission values and gains increasing significance with customer consciousness and competition in the market.

5pEA7. A numerical and experimental investigation into the effects of cavity geometry on the production of cavity tones. Brenda Henderson and Homayun Navaz (Kettering Univ., 1700 W. Third Ave., Flint, MI 48504)

The effect of the leading and trailing edge geometries on the production of tones associated with the flow of low-speed subsonic air over a cavity were investigated experimentally and numerically. The leading edge geometries considered in the study included an overhang (simulating car door gaps) and an overhang with delta tabs such as used to control jet noise. The trailing edge geometries used in the investigations included sloped and rounded impingement regions. The numerical results documented the periodic fluctuations of the transverse velocity, pressure, and density of the shear layer and the resulting cavity pressures. The periodic vortex behavior in the cavity mouth and impingement region was also studied numerically to predict possible configurations for sound suppression. Experimental investigations of the cavity sound-pressure levels were used to determine the validity of the numerical simulations and the predicted cavity configurations for sound suppression.

5pEA8. Hybrid discrete vortex method (DVM)/boundary element method (BEM) calculations for cavity acoustics. Ronald J. Epstein (Phantom Works, The Boeing Co., P.O. Box 516, MC S1067126, St. Louis, MO 63166-0516), Anthony Leonard (California Inst. of Technol., Pasadena, CA 91125), and Alan B. Cain (Phantom Works, The Boeing Co., St. Louis, MO 63166-0516)

A hybrid computational methodology has been developed for cavity acoustics applications. The method utilizes a discrete vortex method coupled to an acoustic boundary element calculation. The discrete vortex method uses a Lagrangian evolution in time and space of a field of Gaussian distributed vortex blobs to simulate a two-dimensional, time-dependent, vorticity dominated shear layer. The method yields accurate and fast unsteady solutions to the time-dependent nonlinear flow equations. The cavity acoustics is modeled using acoustic boundary elements which are distributed on the surface of the cavity geometry. The vortex simulation of the shear layer is coupled to the boundary element calculation through Neumann boundary conditions imposed on the cavity surface. The hybrid method simulates a shear layer interacting with a cavity at low Mach number. In example calculations, acoustic radiation and far-field directivity patterns are calculated for the two-dimensional cavity/shear layer system.

5pEA9. Sound power measurements at low frequencies. Albert Schaffner (Austrian Workers Compensation Board, Adalbert Stifter Str. 65, 1200 Vienna, Austria, albert.schaffner@auva.sozvers.at)

The sound power level of a sound source is usually determined by sound intensity or sound-pressure measurements either in a free-field environment or in enclosures. Although these measurements provide good comparable results for the middle- and high-frequency range, there are large discrepancies at low frequencies. Using the sound intensity method as reference, it can be shown that these discrepancies are systematic. In the case of measuring sound pressure in a free field, they are due to the near-field error depending on the distance of the measurement surface to the source. Measuring sound pressure in enclosures—especially in rooms with hard walls—interference effects cause a change in the sound power output of the source, depending on the distance of the source to the room boundaries. The effects described above have been investigated theoretically and the results have been confirmed by accurate measurements.

5pEA10. Beams of sound: A low-distortion airborne parametric array. F. Joseph Pompei (MIT Media Lab, Cambridge, MA 02139)

An improved airborne parametric array, which uses audio preprocessing and wideband transducers to significantly reduce distortion, is described and demonstrated. The airborne parametric array has been previously shown [Yoneyama *et al.*, J. Acoust. Soc. Am. **73**, 1532–1536 (1983)] to produce highly directional audible sound from the nonlinear self-demodulation of a modulated ultrasonic beam. The resulting audible sound contained a high degree of distortion (up to 50% THD), which is a result of using pure AM modulation and narrow-band transducers. By preprocessing the audio signal and employing wideband transducers, the distortion is shown to be reduced to only 5%, while still maintaining a 3° beamwidth.

5pEA11. Noise problems of the induction motors. Cüneyt Öztürk (Türk Elektrik Endüstrisi AP, R&D Dept., Davutpapa, Litros Yolu, Topkapı 34020, Istanbul, cozturk@tee.com.tr)

The audible noise produced by electrical machines is developed depending on the various energy sources that generally appear during the rotation of the rotor. Usually the main energy source in the electrical machines is the magnetic field. Maxwell stresses act on the stator core, armature, and stator laminations, and generally exhibit spectral distribution as a function of time. These forces may provide excitation on the mechanical resonances of the machine's mechanical system that have a distributed mass, stiffness, and damping. The force waves and their possible contribution to overall machine noise can be predicted by analysis of the electrical design, which can be optimized to reduce their effect. Careful analysis and design are required, with perhaps additional magnetic core material to keep flux levels low, and air gap large, without compromising good performance. Behind the features of the magnetic and mechanical structure, the aerodynamic features and excitations provided by the air circulation and the movement can also gain significance when the similar interaction appears with the machine structure. Interaction might be amplified during the coincidence of the mechanical resonances and the provided excitation on similar frequencies.

5pEA12. The powerful role of the Commanders Decision Aid (CDS) software and acoustic sensor to increasing ground combat vehicle survivability. Ronald M. Yannone (Sanders, a Lockheed Martin Co., 95 Canal St., Nashua, NH 03060)

With the proliferation of diverse anti-tank guided missiles and top attack munitions, ground combat vehicles require the use of multispectral sensors. The issues that arise by adding sensors to these vehicles are the additional cost, size, weight, and power required. Even more critical is the ability of the sensor to provide timely, high-confidence, low-false alarm rate threat detection and classification information to the crew. Amidst a heavy signal clutter battlefield and multiple sensors aboard the vehicle, the concerns are information overload for the crew, being detected by enemy sensors, and quickly depleting limited countermeasure assets by responding to false alarms. The use of infrared and laser warning receivers (IRW, LWR) has been assessed to be very beneficial toward this aim, but not sufficient. Operations analysis has shown that with the addition of an acoustic sensor and decision aid software, crew survivability increases dramatically with low cost. The paper reviews the ground combat vehicle problem space, the data, the complementary data the IRW, LWR, and acoustic sensors provide, and how the Commander's Decision Aid (CDA) software performs multispectral/multisensor fusion, threat classification, threat prioritization, countermeasures response management, and countermeasures effectiveness assessment via various scenario descriptions.

FRIDAY AFTERNOON, 19 MARCH 1999

ROOM H2013, 2:00 TO 5:20 P.M.

Session 5pNSa

Noise: Road Traffic Noise II

Richard F. Schumaker, Cochair
General Motors N.A. Proving Ground, MC 483-356-006 Milford, Michigan 48380, USA

Steven Heinrich, Cochair
TUEV Automotive GmbH, Kaiserstrasse 100, 52134 Herzogenrath-Kohlscheidt, Germany

Invited Papers

2:00

5pNSa1. Perception and meaning of warning signals in urban contexts. Corsin Vogel, Jean-Dominique Polack, Michele Castellengo (Laboratoire d'Acoustique Musicale, CNRS-UPMC-Min. Cult., 11 rue de Lourmel, F-75015 Paris, France, vogel@ccr.jussieu.fr), and Daniele Dubois (CNRS-INaLF, LCPE, 1 rue Maurice Arnoux, F-92120 Montrouge, France)

Perception of warning signals in urban context points out the question of their efficiency with regard to the citizens. Efficiency of warning signals can be described in terms of their meaning for persons who are concerned with. Indeed, subjects can adequately react to the signals only if these latter are identified, that is, only if the signals make sense to the subjects. Therefore, information about mental representations of citizens exposed to warning signals is necessary. An experiment was realized involving 15 signals mixed with two different urban contexts: steady traffic noise and public park ambience. Results show that the two contextual constraints influence the perceptual thresholds of the warning signals as well as their identification. Warning signals can be classified with respect to their different meanings in different urban contexts. Especially, it appeared that signals having an identical semantic description nevertheless may present very different acoustical properties. The classes of signals obtained with the psychological description are a way to determine new acoustical features related to the meaning of warning signals, and so to their efficiency.

2:20

5pNSa2. Application and extension of acoustic holography techniques for tire noise investigations. Fulop Augusztinovicz, János Granszt, Ferenc Mészáros (Tech. Univ. of Budapest, Dept. of Telecommunications, Sztoček u. 2., H-1111 Budapest, Hungary, fulop@hit.bme), Wim Hendrickx, and Herman Van der Auweraer (LMS Intl., B-3001 Leuven, Belgium)

Acoustic holography (AH) is a powerful measurement tool for the investigation of complex sound radiation problems. Enabling detailed source investigations based on noncontact measurements, it is especially useful for the investigation of rotating noise sources such as automotive tires. Nevertheless, conventional AH techniques are burdened by the constraint that the source is assumed to be planar. Based on a detailed measurement series which was performed on a laboratory roller bench, the radiation of a test tire was

investigated by means of AH. It was found that even though valuable information could be obtained, the most important forward and backward radiation of the tire is difficult to tackle in this way. Another approach of the radiation analysis is offered by developing an inverse BEM method. Different implementations of the method have been worked out and tested, based mainly on the SYSNOISE and MATLAB software packages. The technique is applied to experimental tire mock-ups as well as to real-life tire measurements. The practical application and inherent limitations of the technique are also discussed.

2:40

5pNSa3. Measurements of the acoustic absorption coefficient of double-layer asphalts: Analysis of variability. David Casini, Cesare Fagotti, Andrea Poggi, and Simone Secchi (A.R.P.A.T.—Regional Agency for the Environ. Protection of Tuscany, Via San Salvi, 12, 50135, Florence, Italy, arpat.fisica@fi.nettuno.i)

The measurements of the acoustic absorption coefficient were carried out on two kinds of double-layer asphalt located in an urban environment. Three different methods of analysis were used: pseudorandom MLS signal, intensimetry, and Kundt's tube. The measurements with MLS signal were done with a loudspeaker 3-m high and a 1/2-in. microphone 1.5 m above the asphalt. The absorption coefficient has been obtained by the deconvolution of the MLS signal. In the analysis with the Kundt's tube, the transfer function method was used on an asphalt core sample (10-cm diameter) extracted "*in situ*." Two layers, respectively, 2.5 and 5 cm thick, with different granulometry, composed the first asphalt analyzed. In the second asphalt, the superior layer has a larger granulometry. All the results obtained with the different measurement techniques point out a major absorption peak around 630 Hz and a secondary peak around 2000 Hz, for the first asphalt, and 500 and 4000 Hz for the second asphalt. The measurements point out a large variability of the results obtained in different sections of the road. Variation with asphalt's composition and measurement uncertainty is analyzed.

3:00

5pNSa4. Assessment of impact noise determined by binary traffic systems. Miguel A. Sattler (Univ. Federal do Rio Grande do Sul, Dept. de Engenharia Civil, Av. Osvaldo Aranha, 99-3o andar, Porto Alegre, RS, Brazil, CEP 90035-190)

The environmental impact resulting from proposed solutions to solve problems related to the fast growth of Brazilian cities is rarely properly assessed. That is the case of urban traffic noise. There has been a significant growth of vehicles in the last years. In the face of the lack of resources by the public sector to provide adequate public transport for the population, traffic has been increasing considerably year after year. One of the alternatives attempted by the Secretary of Transport of the city of Porto Alegre, south of Brazil, to reduce traffic jams, is to implement an increasing number of binary traffic systems, by using two single-way roads instead of one two-way road. This has increased noise in previously quiet areas of the city. The present paper presents measured values of the resulting noise impact of such measures, considering noise levels before and after the introduction of one of these binary traffic systems.

3:20

5pNSa5. Preventative medical limits for chronic traffic noise exposure. Christian Maschke (Robert Koch-Inst., Federal Inst. for Infectious and Noninfectious Diseases, Section: Environ. Medicine, 12101 Berlin, Germany), Karl Hecht, and Hans-Ulrich Balzer (Inst. for Stress Res. (ISF), 10115 Berlin, Germany)

Because of an expert's report on the medical effects of noise on healthy adults, permissible values for mainly traffic noise have been estimated by the standard of knowledge. If the permissible values are exceeded, then preventative medical action is necessary. Below this value the probability of noise-induced health hazards is essentially zero. The effect due to noise as a health hazard is, besides the mechanical damage of the inner ear, a psycho-physiological deregulation which can be either indirectly due to the annoyance or directly caused by stress of the vegetative-hormonal system. Therefore, different permissible values for annoyance, the stress on the vegetative-hormonal system, and for the loss of hearing are suggested, for both continuous and maximum noise levels. In addition, the deregulation depends on the time of the acoustic exposure because the sensitivity to noise follows a 24-h cycle (circadian rhythm). It is therefore necessary to give personal permissible limits for the evening noise and the nocturnal noise.

3:40–4:00 Break

4:00

5pNSa6. Standardization in the field of traffic noise—An overview of current projects. Ulrich Schober (DIN Deutsches Institut für Normung, D-10772 Berlin, Germany, schober@nals.din.de)

Standardization of vehicle noise measurement procedures is an important instrument for achieving comparable and reproducible measurement results and therefore to provide means for minimizing traffic noise in the long run. In the International Organization for Standardization Technical Committee ISO/TC 43 "Acoustics" and other ISO/TCs many standards about noise emission measurements on all kinds of vehicles including aircraft, and about tire/road noise effects and road surface characteristics are currently under preparation in several working groups. Some standards are under revision to bring them in line with the advances in vehicle technology, other standards are prepared due to the needs from vehicle manufacturers and environmental protection representatives. The lecture presents ongoing developments in important projects and emphasizes their joint aspects and differences. Examples: (i) Revision of ISO 362 about noise measurement on accelerating cars and motorcycles which is important for type testing. (ii) Project about measurement of tire noise. (iii) Project about the influence of road surfaces on traffic noise to facilitate acoustical characterization of road surfaces (ISO 11819 series). (iv) Revision of ISO 3891 about aircraft noise. (v) Projects about noise emission measurements on vessels (revision of ISO 2922) and small recreational craft (new ISO 14509). (vi) Revision of ISO 3095 about noise emission of railbound vehicles.

5pNSa7. Cadastre of the acoustic characteristics of the whole road network of the Brussels region. Jean-Pierre Clairbois, Peter Houtave (Acoustical Technologies, Av. Brugmann 215, B-1050 Brussels, Belgium), Andreas Koellmann (TUV-Automotive, D-52134 Herzogenrath, Germany), and Gerhard Mosdzianowski (B-4721 Kelmis, Belgium)

The Brussels Institute of the Environment (IBGE-BIM) took the initiative to start a global study assembling elements of its future noise planning. Road surface noise is an important element of these. A strong methodology has been decided in order to get an exhaustive database of the whole road network of the Brussels region (1800 km). The idea is to get an interactive tool in order to control the actual conditions of the road surfaces and their influence on the global road traffic noise, and also to be able to look at the different possible improvement scenarios. Establishing this database required not only to visit every road of the network, but also to assemble all relevant data for estimating the road and traffic conditions. The methodology and tools used for the work, the results, and a statistical analysis of the database are presented. This work has been completed by noise measurements (CPB and trailer methods) of the nine different kinds of road surfaces used in the Brussels region. A strategy of action is suggested and the maintenance of the database is started so that the evolution of the network can be continuously considered within the noise planning. [Work supported by IBGE-BIM.]

5pNSa8. Modeling the propagation of traffic noise in topographically modified atmospheric environments. Dietrich Heimann (DLR, Institut für Physik der Atmosphäre, Oberpfaffenhofen, D-822234 Wessling, Germany, d.heimann@dlr.de)

A meteorological mesoscale model is used to simulate the time-dependent distribution of temperature and the corresponding development of autochthonous slope-wind systems in a narrow two-dimensional valley during the course of a cloud-free day. Further simulations consider the topographical modification of allochthonous airflow across the valley. A numerical sound particle model was designed to cope with both the uneven terrain and the inhomogeneous meteorological environment. The acoustical model takes up the simulated meteorological fields and calculates the propagation of noise which originates from a steady source (road or railway) in the valley. The coupled modeling system ensures consistency of topography, meteorological parameters, and the sound field. The temporal behavior of the sound-pressure level across the valley is examined. The results show remarkable, meteorologically induced variations of the sound level during the course of a cloud-free day. In addition, it is shown how the location of a planned traffic way can be optimized with respect to minimum noise impact in the valley [D. Heimann and G. Gross, Appl. Acoust. (in press)].

5pNSa9. The effect of road features on the traffic noise—Prediction and measurement. Wolfgang Foken (Westfälische Hochschule Zwickau (FH), Kraftfahrzeugtechnik, Postfach 201037, D-08012 Zwickau, Germany, wolfgang.foken@fh-zwickau.de)

In order to calculate the noise level from road traffic, prediction methods are used in most countries, for instance the “RLS90” (“Guideline for Noise Control at Roads”) is the relevant calculating method in Germany. The initial situation is always the amount of cars, the percentage of trucks, and their velocities. In addition, features of the road are required, for example, the road pavement, the gradient, and the existence of traffic lights. They are considered to be surcharges. The aim of this paper is the critical evaluation of these road features and the comparison with measured data. The paper shows that traffic lights generally do not cause an increase of the noise level. It was noted that the surcharge for traffic lights depends on the traffic density and the percentage of trucks. Therefore, a general usage of the “RLS90” surcharge for traffic lights is questionable. The German “RLS90” often underestimates the effect of the road pavement. Measurements also indicate the necessity of defining the surcharge for the road pavement as a continuous function of the velocity. The surcharge for the road gradient does not depend on the velocity but a reliance on the percentage of trucks is assumed.

Contributed Poster

This poster will be on display in the Poster Gallery from Thursday to Friday, 18–19 March. The author will be at the poster from 2:00 p.m. to 4:00 p.m. on Thursday, 18 March.

5pNSa10. Effect of the acceleration on vehicle noise emission. Joel Lelong and Roger Michelet (Dept. MMA, Inrets, 25 av. F. Mitterrand, Case 24 69675, Bron Cedex, France)

In urban areas, traffic noise, considered as one of the most important public nuisances, is generated by stop-go traffic. Irregular urban traffic is directly linked to traffic regulation devices, which are the cause of frequent accelerations/decelerations of vehicles, depending on driving behavior. In order to predict sound levels at the time of conception, traffic regulation devices, measurements of acceleration passby noise, are necessary. Urban streets are synonymous with complex acoustical environments (presence of reverberant obstacles and multiple sources of noise): carrying

out of *in situ* vehicle noise emissions measurements is impossible. Measuring vehicles' noise emissions on test tracks can overcome this difficulty, therefore resulting in a better understanding of the acceleration effect on vehicle noise emission. The method described in this paper is based on simultaneous measurements of kinematical data (speeds, accelerations), mechanical parameters (gear ratio, engine speed), and acoustical levels ($L_{A\max}$). First results obtained on cars show significant differences between acceleration levels and cruise levels obtained at same speeds. The level of increase depends on the acceleration value, the gear ratio, and the engine speed. These first results have been compared to existing published results. [Work supported by French Research Program on Ground Transportation (PREDIT).]

Session 5pNSb**Noise: Future Noise Policy in Europe**

Tjeert ten Wolde, Cochair
European Commission, Rue de la Loi 200, B-1049 Brussels, Belgium

Paul D. Schomer, Cochair
USA CERL, P.O. Box 9005, Champaign, Illinois 61826-9005, USA

Invited Papers**2:00**

5pNSb1. The new noise policy of the European Union. Tjeert ten Wolde (European Commission, Rue de la Loi 200 (TRMF 1/68), B-1049 Bruxelles, Belgium, Tjeert.Tenwolde@dg11.cec.be)

In 1996 the European Commission published a so-called Green Paper in which proposals were done for a future EU policy on environmental noise. More than 450 institutions and individuals commented and on the basis of these the European Commission concluded that there is more than sufficient support for most of the proposals and started the process of the further development of the policy. Central elements in the policy are the harmonization of the noise indicators and the assessment methods in the European Union, the introduction of noise mapping and action plans for cities and member states, and information of the public and periodical EU action plans. Target setting will probably remain the responsibility of the Member States or local authorities. It is the aim that the coherence and quality of the EU policy on noise sources will improve. In 1998 working groups started with the development of indicators, dose-effect relationships, computation and measurement methods, mapping procedures, and guidelines for abatement plans. An overview of their tasks and their progress shall be presented. In parallel the development of a first Directive for environmental noise began. The progress with this process shall also be reported.

2:20

5pNSb2. A European noise indicator. Martin van den Berg (Ministry of Environment, Noise Directorate IPC 635, pbox 30945, 2500 GX Den Haag, The Netherlands)

The European Commission installed five working groups with tasks to advise the Commission on a future European noise policy. The first working group, chaired by the author, was required to advise on European noise indicator. From research that was carried by INRETS [study related to the preparation of a communication on a future EC noise policy, LEN Report nr 9420 Lyon (1994)] it was clear that a large variety exists in noise indicators and calculation methods, which makes it difficult to compare efficiency of abatement measures. According to the time schedule the working group is to produce its final report in February 1999. In the presentation the conclusions of the working group will be discussed.

2:40

5pNSb3. EU's future noise policy: Progress report of WG Dose/Effect. Jacques Lambert (INRETS, Case 24, 69675 Bron Cedex, France, lambert@inrets.fr)

In order to assist the preparation of the EU's future noise policy, WG 2 (Dose/Effect) has launched a work program that will: provide dose/effect relationships for annoyance, sleep disturbance, speech interference (and other effects if necessary), provide proposals for noise indices, define needs on dose/effect research and play an active role in the initiation and coordination of that research, make proposals for the dissemination of information to the public and to authorities (national, local and regional), and make a proposal for the setting of targets by authorities making use of harmonised dose/effect relationships. An inventory of existing material is now available as well as an identification of research needs (March 1999). A position paper will be delivered to the European Commission by 2002. It will focus on thresholds and dose-effect relationships related to transportation noise (road, rail, and aircraft), industrial noise, entertainment and recreational noise, on the setting of targets for the point of view of health effects, and dissemination of gathered information.

3:00

5pNSb4. Harmonization of computational and measurement methods for noise exposure in the European Union. Dieter P. A. Gottlob (Federal Environ. Agency, Bismarckplatz 1, D-14193 Berlin, Germany, d.u.gottlob@t-online.de)

In the framework of the new European noise policy, the European Commission has created working groups to assist in the preparation of this policy. The members are appointed by the Commission on the recommendation of the member states, local authorities, and various organizations. Working group 3 (WG 3) is dealing with the harmonization of calculation and measurement procedures for noise assessment, mapping, planning, and noise abatement. Calculation methods shall be elaborated for road, rail, aircraft, outdoor machinery, and industry. They shall be suited for calculating the long-term noise exposure as well as single events, especially from air and rail traffic. Additionally, they shall be developed for a variety of geometrical and weather conditions which occur in the member states. At the Copenhagen Conference (September 1998), WG 3 started working and discussed the work program

with experts from various countries. It was decided that emission data and transmission models should be separated. Emission measurement procedures should be provided, and the propagation models currently in use should be improved, especially in their physical representation. In the next months an action plan will be elaborated. In this paper the current state of the discussions will be presented.

3:20

5pNSb5. A progress report on E. C. proposed noise mapping. John Hinton (Environ. Services Dept., Birmingham City Council, England, bham.acoustics@dial.pipex.com)

As an integral part of the development of a framework directive for the assessment and reduction of environmental noise, the European Commission has set up five working groups on the perception of noise. Working Group 4 (W.G.4) is investigating noise mapping techniques and is required to produce a position paper for the E. C. giving guidelines for effective noise mapping. It is now widely accepted that the provisions of mapping information is essential to a process designed to reduce exposure of citizens throughout Europe to unacceptable levels of environmental noise. This will be achieved by providing the public and politicians with information on the scale of the problems that are faced and by providing decision makers at local, regional, national, and European level with the information they require to develop action plans for noise reduction. Mapping will also provide a means of monitoring the implementation and effectiveness of such plans. W.G.4 held its first meeting just prior to the Invitational Conference on the European Unions Future Noise Policy held in Copenhagen in early September 1998. The paper will provide a progress report of W.G.4.

3:40–4:00 Break

4:00

5pNSb6. A new EU directive concerning the noise emission by equipment used outdoors. Volker K. P. Irmer (Umweltbundesamt, Berlin, Germany)

The Green Paper "Future Noise Policy" [COM(96) 540 final] identifies noise in the environment as one of the main environmental problems in Europe. It emphasizes the continuing need for legislative measures at the Community level concerning noise emission from various sources. In February 1998, the Commission adopted a proposal for a European Parliament and Council Directive on the approximation of the laws of the member states relating to the noise emission by equipment used outdoors [COM(1998) 46 final, published in OJ No. C 124, 22.04.1998, p. 1]. It intends to simplify legislation, to contribute to the smooth functioning of the internal market, and to protect the health and well-being of citizens by reducing the overall noise exposure. The proposal covers 55 types of equipment, all of which are to be marked with the guaranteed noise-emission level; in addition, noise limits are laid down for 19 types of equipment (13 already covered by existing directives, 6 new ones). Noise limits are laid down in two stages (the first one eliminating most noisy equipment, the second one taking into account advanced technology). The requirements of the proposal are explained and the progress made in the Council and European Parliament is reported.

4:20

5pNSb7. The action programme of UIC and CER (Community of European Railways) "abatement of railway noise emissions from goods trains." Peter Hübner (Deputy Director Way and Works, President of UIC-Task Force Noise & Vib., Swiss Federal Railways, Mittelstrasse 43, CH-3030 Berne, Switzerland marianne.mb.binggeli@sbb.ch)

Although rail transport has a low environmental impact overall, noise from goods trains remains a major problem. Research has identified that wheel roughness is the critical factor and that composite materials will deliver adequate braking with less damage to the wheel surface than existing iron brake blocks. Accordingly the UIC/CER has made a formal commitment to fit composite blocks to existing wagons, as well as to all new wagons. Existing wheels cannot cope with the thermal stress from composite blocks. Fitting new wheels only when the old are worn out will be cost neutral, but will take 15–20 years. Premature wheel replacement will involve additional cost, but will deliver the benefits in 5 years. This can be achieved with financial assistance from E.U. member-states, who will thereby avoid unnecessary outlay on noise barriers. At the same time proposals for a E.U. noise creation standard should reflect the performance achieved by the modified wagons. The railways propose a voluntary environmental agreement with the E.U.

Contributed Paper

4:40

5pNSb8. Characterization of quiet areas: Subjective evaluation and sound-level indices. Dick B. Botteldooren, Saskia Decloedt, and Jürgen Bruyneel (Dept. of Information Technol., Univ. of Gent, St. Pietersnieuwstraat 41, 9000 Gent, Belgium)

In densely populated regions, a government may decide to conserve quiet areas within reach of the average population. To characterize such areas, long-term exposure measures and typical limits have to be decided

upon. Since the main goal of the silent area is to provide repose for Man, a subjective evaluation of silence may also be envisaged. In this paper, subjective evaluation of silence and sound-level measurements obtained in two related, but independent studies of open area are compared. Thorough statistical analyses lead to the conclusion that statistical noise levels with larger index (L_{A95} to L_{A50}) are best suited to predict subjective silence. Critical levels are obtained. They lead to over 80% correct classification for both studies. Reported disturbance, on the other hand, seems correlated to events, but does not have a strong influence on the global rating of silence.

Session 5pPAa

Physical Acoustics: Acoustic Remote Sensing of the Atmosphere II

D. Keith Wilson, Cochair

U.S. Army Research Laboratory, 2800 Powder Mill Road, Adelphi, Maryland 20783, USA

Volker Mellert, Cochair

Physics Department, University of Oldenburg, D-26111 Oldenburg, Germany

Invited Papers

2:00

5pPAa1. Acoustic tomography as a remote sensing method inside the atmospheric surface layer. Astrid Ziemann, Klaus Arnold, and Armin Raabe (Universität Leipzig, Institut für Meteorologie, Stephanstr. 3, D-04103 Leipzig, Germany, ziemann@rz.uni-leipzig.de)

Acoustic travel time tomography will be applied on the atmosphere to directly provide spatially averaged values of meteorological quantities (temperature and wind field) which are needed for the evaluation of numerical models as well as to complete *in situ* point measurements and conventional studies of the atmospheric surface layer over natural surfaces. Sound waves propagate through the atmosphere with different sound velocities according to the distribution of air temperature and wind vector and become, therefore, an information carrier about the medium properties. In the presented study acoustic travel time data were measured between sources and receivers placed around an array of $200 \times 260 \text{ m}^2$ as initial values for the tomographic reconstruction. Thereby, a frequently applied procedure of ray tomography, the simultaneous iterative reconstruction technique, characterized by simple handling and small computational requirements, was used. Additionally, a special system of equations for the isolation of the temperature and wind influence on the sound speed was solved. To estimate the sound ray propagation and to simulate acoustic travel times during various meteorological conditions a ray tracing model based on a modified version of Snell's law corresponding to the influence of wind vector will be introduced.

2:20

5pPAa2. Opto-acoustic sounding of the atmospheric parameters. Lyudmila G. Shamanaeva (Inst. of Atmospheric Optics of the SB RAS, 1, Akademicheskii Ave., Tomsk 634055, Russia, zuev@iao.tomsk.su)

Experimental investigations of sound pulse generation accompanying the propagation of high-power pulsed laser radiation in the atmosphere were started at the IAO SB RAS in 1986. Based on these investigations, new opto-acoustic methods of sounding of the atmospheric temperature, wind velocity, relative humidity, and number density of micron fraction of atmospheric aerosols were suggested. A source of the acoustic signal in an opto-acoustic sounding system is a discrete laser spark made by focusing a CO_2 laser beam into the atmosphere at distances up to 500 m. The number density of coarsely dispersed aerosols was determined from the number of pulses in the acoustic signal generated by the discrete laser spark. Results of opto-acoustic measurements of the number density of coarsely dispersed aerosols agree well with the microphysical model of aerosol atmosphere for Western Siberia. A threshold laser energy density resulting in the origin of local plasma formations is about $6\text{--}10 \text{ J/cm}^2$. For lower laser energy densities, the acoustic radiation is generated due to laser-induced thermal expansion of the propagation medium. The measurable parameters are the effective laser beam radius, the coefficient of laser radiation absorption, and the total laser beam energy.

2:40

5pPAa3. Radio acoustic sounding of the planetary boundary layer. Gerhard Peters (Meteorological Inst., Hamburg Univ., Bundesstrasse 55, D20146 Hamburg, Germany)

Refractive index variations of the atmosphere caused by sound waves are used as tracers for radar waves in order to determine the sound velocity and thus the (virtual) temperature remotely. This combined usage of acoustic and electromagnetic waves—called “radio acoustic sounding system (RASS)” —has gained considerable interest, as sensitive radar wind profilers have been used for one decade in atmospheric research and monitoring. The RASS function can easily be added to radar wind profilers with the help of sound sources at the radar site. Due to the sound attenuation, the range of most systems is limited to the typical depth of the planetary boundary layer, which is particularly important for the exchange of mass, energy, and momentum between atmosphere and surface. Due to refinements of the signal analysis, a measurement accuracy comparable to conventional *in situ* sensors has recently been achieved. In contrast to *in situ* measurements, RASS has the great advantage to provide continuous and simultaneous data from many altitudes. RASS cannot only deliver accurate temperature profiles but also profiles of the mean wind and of high-frequency fluctuations of the vertical wind component. Currently a RASS-based method is being developed for high-resolution horizontal wind measurements.

5pPAa4. The application of SODAR and RASS configurations in operational meteorology. D. Engelbart (Deutscher Wetterdienst, Meteorol. Observatorium, D-15864 Lindenberg, Germany, engelbart@mol.dwd.d400.de)

In operational meteorology, the recent improvements in scale resolution as well as physical parametrizations of numerical weather prediction models imply the demand of improvements in its measured data base, i.e., in measuring systems. Compared with the classical aerological sounding by radiosondes, supplying data of meteorological standard parameters twice per day, active remote-sensing systems like SODAR and RASS are able to increase both time resolution and vertical resolution considerably depending on system configuration. By using pure acoustical sounding a SODAR allows the derivation of vertical profiles of the 3-D wind vector, whereas integrated systems like SODAR and RASS or windprofiler and RASS are able to supply temperature profiles as well. A crucial point regarding assimilation of operationally measured data into numerical models is described by data quality. A second general point is focused on the availability of remotely sensed data. Generally, in operational meteorology acoustic remote sensing may aid in three major tasks, i.e., data supply of operational numerical models, improvement of physical parametrizations by supplying validation data and monitoring of complex parameters, e.g., boundary-layer height. As an example for model validation the presentation finally will show some results of SODAR- and RASS-derived flux profiles in comparison to high-resolution model results.

3:20

5pPAa5. Acoustic sounding of fog and rain. Sergei V. Shamanaeva and Lyudmila G. Shamanaeva (Inst. of Atmospheric Optics of the SB RAS, 1, Akademicheskii Ave., Tomsk 634055, Russia, zuev@iao.tomsk.su)

The coefficients and phase functions of sound scattering by a single rigid spherical particle whose parameter Mie changes in a wide range have been calculated with the use of exact formulas of the Mie scattering theory. Analytical expressions are given for the dependence of sound attenuation on the parameters of fog and rain and acoustic frequency. The acoustic scattering cross section of a single particle has been derived. New methods of acoustic sounding of fog and rain have been suggested and theoretically analyzed. Experimental data on the transformation of raindrop size distribution have been obtained by processing of raw spectra of acoustic signals recorded in rain for 20 s and 1, 2, 3, 5, and 25 min with cw bistatic sodars operating at frequencies of 5 and 40 kHz. The bimodal character of the raindrop size spectra was established. The position of the second maximum in the raindrop size spectrum was shifted with time toward larger raindrop sizes, from 0.3 to 1 mm. The estimated rain intensity varied from 0.5 to 1.9 mm per hour. The obtained results demonstrate that high-frequency cw bistatic sodars are very promising for measurements of fog and rain.

3:40

5pPAa6. Acoustic pulse sounding of the atmospheric boundary layer. Igor P. Chunchuzov (Obukhov Inst. of Atmos. Phys., Russian Acad. of Sci., Pyzhevskii 3, Moscow 109017, Russia)

The method of acoustic pulse sounding of the atmospheric boundary layer is used to study spatial coherence and frequency spectra of the pulse travel time fluctuations between the pulse source and the triangular set of receivers 2.5 km apart from each other. The wind speed fluctuations were measured up to 200–300 m with the use of SODAR and the anemometers placed on the mast of 56-m height. The maximum coherence between the wind speed and pulse travel time fluctuations was obtained for the period of 8 min. This period is thought to be associated with the ducted internal waves in the troposphere. The use of the pulse sounding for the acoustic tomography of the atmospheric boundary layer is proposed.

4:00–4:20 Break

Contributed Papers

4:20

5pPAa7. Acoustic investigation of a vorticity filament using time-reversal mirrors. Sébastien Manneville, Agnès Maurel, Mathias Fink (Lab. Ondes et Acoust., E.S.P.C.I., Univ. Paris VII, URA CNRS 1503-10 rue Vauquelin, 75005 Paris, France, sebastien.manneville@espci.fr), Frédéric Bottausci, Philippe Petitjeans, and Eduardo Weisfred (E.S.P.C.I., URA CNRS 75005 Paris, France)

A new experiment of a filamentary vortex in water characterized by ultrasound is presented. The flow is generated by two corotating discs in an infinite medium, and suction can be applied at the center of the discs. Acoustic time reversal mirrors [see Roux *et al.*, ASA Proc. 1433 (1998)] provide dynamical, nonintrusive, and global measurements of the vortex characteristics. Three different regimes are observed depending on the control parameters (distance between the discs, rotation frequency of the discs, and suction flow rate): (1) a regime of stable, stationary vortex where the velocity profile is compared to the Burgers model and scaling laws are investigated when varying the control parameters, (2) a regime of intermittency with cycles of vortex breakdowns, and (3) a domain where the vortex never shows up.

4:40

5pPAa8. Investigation of the effects of acoustic refraction on Doppler measurements caused by wind- and temperature profiles. Annette Schomburg, Detlef English, and Volker Mellert (Dept. of Phys., Acoust., C.v.O., Univ. of Oldenburg, 26111 Oldenburg, Germany)

The subject of this investigation is the errors appearing in Doppler SODAR measurements due to wind and temperature profiles. Usually the wind velocity is calculated from the measured Doppler shift under the assumption of straight line propagation. Wind and temperature profiles cause a deviation of the rays from the straight path which is given by the choice of geometry between transmitting antenna, receiving microphone and scattering volume. The appearing errors in the magnitude of the Doppler shift because of the displacement of the scattering volume, changes of scattering angle, and scattering vector have been investigated. Results from simulations, where the exact position of the scatterer, as well as the ray deformation, has been taken into account, show that upwind and downwind errors are symmetrical. This leads to a new method of measuring wind profiles. Outdoor measurements performed with a new SODAR configuration will be presented. Four receivers allow a construction in which each receiver pair builds a plane to measure down- and upwind components. The resulting two wind profiles describe the horizontal wind

component. With a fifth receiver near the source, the vertical component should be measurable, so that the complete wind vector and even the temperature profile could be determined.

5:00

5pAa9. Sodar-derived vertical profiles of the parameters of atmospheric turbulence. Lyudmila G. Shamanaeva and Aleksandr Ya. Bogushevich (Inst. of Atmospheric Optics of the SB RAS, 1, Akademicheskii Ave., Tomsk 634055, Russia, zuev@iao.tomsk.su)

The vertical profiles of the structural characteristics of temperature and wind velocity fields and longitudinal and transverse structure functions of the wind velocity field measured with the Zvuk-2 three-channel Doppler sodar are presented for altitudes of 75–535 m above the ground with a 20-m altitude resolution. The sodar operated at a frequency of 1700 Hz, its pulse repetition period was 10.5 s, and its pulse length was 150 ms. One transceiving antenna was pointed vertically and two others were tilted at angles of 20 degrees from the vertical in orthogonal planes. Backscattered signal power was additionally measured. Sodar data were interpreted using the iterative algorithm considering the excess turbulent attenuation of a sodar signal. Sodar-derived structural characteristics were compared with simultaneous measurements by an ultrasonic anemometer/thermometer. The structure functions of the wind velocity field were calculated from long series of observations of vertical profiles of the wind velocity vector. At night when the wind shear was observed the longitudinal structure functions below it were strongly suppressed, whereas above the wind shear they sharply increased. Under conditions of daytime convection the structure functions first increased with the separation between the observation points and then saturated.

5:20

5pAa10. The Radar-RASS spaced-antenna wind profiler, a tool to measure atmospheric turbulence. Lutz Hirsch (Max-Planck-Inst. for Meteorol., Bundesstr. 55, 20146 Hamburg, Germany, hirsch@dkrz.de)

Recently a Radar-RASS spaced-antenna wind profiling system with four receiving antennas has been developed. The system was designed to use the RASS-signal for radar-interferometry measurements of high-resolution time-series of the 3-D wind vector. The use of the deterministic

RASS-backscatter signal which is insensitive to clutter interferences from the ground or elevated targets makes this system a useful tool for turbulence observation in the atmospheric boundary layer. The well-known problem of overestimating the horizontal velocity using simple correlation algorithms, caused by decaying refraction pattern, had to be addressed carefully. Multiple published approaches to find the correct horizontal velocity in the frequency domain using power and cross spectra as well as in the time-domain using correlation-functions were compared to each other. Based on these comparisons the implementation of the most suitable algorithm for the present system together with problems like noise suppression or system constants estimation will be discussed. Results of field measurements will be presented and the performance of the system to derive high-resolution time series of the 3-D wind vector will be demonstrated. Average quantities will be compared with the results of a SODAR operated next to the RASS.

5:40

5pAa11. Experimental method of acoustic tomography of the atmospheric surface layer. Klaus Arnold, Astrid Ziemann, and Armin Raabe (Institut für Meteorologie, Universität Leipzig, Stephanstr. 3, 3-04103 Leipzig, Germany, arnold@rz.uni-leipzig.de)

Acoustic tomography is proposed as a technique for remote monitoring of near surface temperature and wind fields. An experimental method is presented which provides area-averaged values of the demanded meteorological parameters. In this study travel time tomography was used by measuring the travel time of a signal at a defined propagation path. In such a system a number of sources simultaneously transmit an acoustic signal (sine oscillation) which is detected at a number of receivers. The travel time of each signal was estimated by cross correlation between the received and the transmitted signal. Each peak of the cross correlation is associated with a separate ray path and the delay time corresponds to the travel time of the transmitted signal. The travel time data were inverted to obtain estimates of the meteorological parameters. A field experiment was carried out in autumn 1997 at the test site Melpitz near Leipzig to provide the input data for the tomographic reconstruction. Six sources and four receivers were positioned at an array of 200×260 m². Derivations of area-averaged values result from inversion of the single values of travel time for all ray paths.

FRIDAY AFTERNOON, 19 MARCH 1999

ROOM MA042, 1:55 TO 5:20 P.M.

Session 5pAb

Physical Acoustics: Sonochemistry

J. Reisse, Cochair

Sciences Appliquées, Université Libre de Bruxelles, Av. F. D. Roosevelt 50, B1015 Bruxelles, Belgium

Kenneth S. Suslick, Cochair

Department of Chemistry, University of Illinois, 600 South Mathews Avenue, Urbana, Illinois 61801, USA

Chair's Introduction—1:55

Invited Papers

2:00

5pAb1. Sonochemical synthesis of new materials. Kenneth S. Suslick, Gennady Dantsin, Arash Ekhtiarzadeh, and Arul Dhas (Univ. of Illinois, 600 S. Mathews Ave., Urbana, IL 61801, ksuslick@uiuc.edu)

High-intensity ultrasound has found new applications in the synthesis of unusual inorganic materials. The chemical effects of ultrasound originate from acoustic cavitation, which produces local transient conditions of ~ 5000 K, ~ 500 atm, with heating and cooling rates that exceed 1×10^{10} K/s. In cold liquids, ultrasound is able to drive reactions that normally occur only under extreme

conditions. Examples include intercalation, activation of liquid–solid reactions, and synthesis of amorphous and nanophase materials. The sonochemical syntheses of nanophase metals, alloys, metal carbides, supported heterogeneous catalysts, and nano-colloids derives from the sonochemical decomposition of volatile organometallic precursors during cavitation, which produces clusters a few nm in diameter. Such nanostructured solids are active heterogeneous catalysts for various reactions. Most recently, we have discovered a new synthesis of nanometer-sized MoS₂ and of transition metal colloids of iron, cobalt, and iron–cobalt alloy colloids. Sonication of molybdenum hexacarbonyl in the presence of sulfur produces a novel morphology of MoS₂ with extremely high activity for catalytic hydrodesulfurization of thiophene. Stabilized iron, cobalt, and alloy colloids have also been prepared that are amorphous as initially prepared with particle sizes ~5 nm as determined by TEM. Magnetic studies show that the colloids are superparamagnetic and function as useful ferrofluids.

2:20

5pPAb2. On the frequency effect and the solvent effect in sonochemistry. Kristin Bartik, Nicolas Segebarth, Juliana Vandercammen, and Jacques Reisse (Lab. de Chimie Organique E.P., Univ. Libre de Bruxelles, 50 av. F. D. Roosevelt, 1050 Bruxelles, Belgium)

The frequency effect in sonochemistry remains an open question. On the basis of data reported in the literature it seems that sonochemistry in water is faster in the high-frequency range (between 500 kHz and 1 MHz) than in the low-frequency range (between 20 and 100 kHz). This work is devoted to the study of the frequency effect in various solvents. The study of a frequency effect is far from trivial. Indeed, a systematic study would require that all the other parameters of the system be kept constant. This condition is never fully fulfilled. Heat dissipation and electrical energy consumption were used to characterize the ultrasound intensity even if criticisms can be formulated against these very indirect methods of measuring the ultrasound intensity. Various sonochemical reactions were studied. The frequency change (with all that it encompasses) clearly has an effect, and this effect is different in water, organic solvents, and emulsions. An interpretation of these observations will be proposed.

2:40

5pPAb3. Production of free radicals in ultrasound fields. Gareth J. Price (Dept. of Chemistry, Univ. of Bath, Bath BA2 7AY, UK, g.j.price@bath.ac.uk)

It is well known that a major consequence of cavitation collapse in liquids is the production of highly reactive chemical species such as free radicals. These can be used to perform useful chemistry, although in some cases they can be undesirable byproducts of a reaction. A series of trapping techniques to study radical formation in organic and aqueous systems has been developed. These include the use of UV/visible, fluorescence, and electron spin resonance spectroscopies. The paper will describe the methods and give some results to demonstrate the effects of varying the experimental conditions on radical formation. Examples will be taken from synthetic chemistry, from polymerization reactions, and from aqueous solutions (including biomedical ultrasound sources) to illustrate the factors of importance and the implications of the results for these processes.

3:00

5pPAb4. Sonolysis of nitrophenol and nitrophenyl acetate in aqueous solution. C. von Sonntag, A. Tauber, and H.-P. Schuchmann (Max-Planck-Institut für Strahlenchemie, P.O. Box 101365, D-45413 Mülheim, Germany, vonsonnt@mpi-muelheim.mpg.de)

The dual aspects of aqueous sonochemistry can be nicely demonstrated by the use of nitrophenol as a substrate. The nature and sonolytic yield of the products depends drastically on the state of dissociation of this substrate (pK_a 7.1). At pH 4, the undissociated compound, being hydrophobic, is enriched in the boundary layer of the cavitation bubble and develops a partial pressure within the bubble, in contrast to the situation at pH 10 where the phenolate anion exists. Accordingly, at pH 4 the degradation is gas-phase pyrolytic, but at pH 10 it occurs in the aqueous phase induced by OH radicals. There has been a debate as to whether thermohydrolysis of substrates such as 4-nitrophenyl acetate, which has been thought to proceed in a layer of “supercritical” water surrounding the collapsing cavitation bubble, plays a role in their aqueous-solution sonolysis. New results, especially the absence of nitrophenol formation, indicate that this is not the case. This, as well as the formation of considerable amounts of OH-radical-induced and “deep pyrolysis” products clearly shows that the 4-nitrophenyl acetate system is incapable of specifically probing any domain of supercriticality.

3:20

5pPAb5. Sonoluminescence from water–ethylene glycol mixtures. Melanie Bradley, Muthupandian Ashokkumar, and Franz Grieser (Adv. Minerals Products Res. Ctr., School of Chemistry, Univ. of Melbourne, Parkville, 3052 Australia)

The emission obtained from air-saturated ethylene glycol–water mixtures has been examined using 3-ms pulses of 500-kHz ultrasound. In pure ethylene glycol at 24 °C the intensity of the signal obtained is about eight times weaker than that in pure water. At 50/50 vol.% the signals are comparable to pure water. Addition of aliphatic alcohols to a 50/50 vol.% solution quenches the signal, and the more hydrophobic the alcohol, the greater the effect. Addition of the anionic surfactant sodium dodecylsulfate enhances the signal at low concentration (1–2 mM). As the percentage of ethylene glycol in the mixture increases, the extent of the alcohol quenching and the surfactant enhancement is diminished. The effect of the additives can be explained in terms of their absorption at the solution/bubble interface and their role in controlling bubble-bubble coalescence. In addition for the case of alcohol quenching, it is postulated that evaporation into the bubble also occurs and the presence of this alcohol in the core of the bubble interferes with the processes that produce sonoluminescence. This latter conclusion is similar to what we have previously raised for sonoluminescence quenching in water by alcohols [Ashokkumar *et al.*, J. Phys. Chem. **101**, 10845–10850 (1997)].

5pPab6. Bridging the gap between single- and multibubble sonoluminescence. Thomas J. Matula (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

Sonoluminescence from a single bubble (SBSL) is related to sonoluminescence from a cavitation field (MBSL) by a commonality of cause (acoustic cavitation) and effect (light emission). However, questions remain as to the extent of this association. Furthermore, detailed comparisons are difficult due to the dissimilar approaches used in generating, and conditions amenable to, sonoluminescence from a single bubble versus a cavitation field. Previous comparisons of the spectral and temporal characteristics of the light emission have been limited by the sparse quantity of available data [T. J. Matula and R. A. Roy, "Comparisons of sonoluminescence from single-bubbles and cavitation fields: Bridging the gap," *Ultrasonics Sonochemistry* **4**, 61–64 (1996)]. Further comparison studies are currently underway in order to establish whether or not a more direct relationship exists between MBSL and SBSL. These studies include a near-IR spectral comparison, a comparison of the effects of doping with small quantities of impurities, and a comparison of the effects of an overpressure on the light intensity. An increased understanding of the relationship between these two phenomena should be extremely useful to sonochemistry researchers. [Work supported by DOE-EM.]

4:00–4:20 Break

Contributed Papers

4:20

5pPab7. Ultrasonic generator with frequency modulation used in sonochemistry. Mircea Alexandru Sarbu, Elek Demeter (Res. Inst. for Electron. Machines, Bd. Tudor Vladimirescu 45, 79623 Bucharest, Romania, sarbuma@pcnet.pcnet.ro), Mircea Vinatoru (Inst. of Organic Chemistry, 71141 Bucharest, 15-254, Romania), and Adrian Daniel Constantinescu (Res. Inst. for Electrotechnics, Splaiul Unirii 313, 74204 Bucharest, Romania)

The electronic generator includes an oscillator and a power stage working in tandem. It is supplied with an unfiltered rectified tension voltage. The signal applied to the transducer is completely modulated in amplitude, having a modulation sinusoidal frequency of 100 Hz. The adaptation transfer of power is realized by a voltage transformer, in this case the rate of voltages being 3.2. The input power was halved by a separate connection, at which one gets 0.707 of the maximum tension voltage. The output signal of the oscillator is frequency modulated with a white noise signal, the resulting signal having a frequency deviation less than 10% of the oscillation frequency. Frequency modulation by a white noise signal generates a broadband signal, very useful for the improvement of the efficiency of the chemical reaction. The carried experiments proved that the use of half of the transmitted power led to interesting results, too. So, for the case of the Weissler reaction of potassium iodine decomposition, at the same time of sonification there was practically a double concentration of the obtained substance, iodine, for the case of half of the power generator.

4:40

5pPab8. Effect of crosslinking on dynamic mechanical properties of polyurethanes. Jeffrey J. Fedderly, Gilbert F. Lee, and John D. Lee (Carderock Div. Naval Surface Warfare Ctr., West Bethesda, MD 20817-5700, leejd@dt.navy.mil)

The dynamic mechanical properties of five polyurethane compositions of varying crosslink density were measured using a resonance technique. The five crosslinked polyurethanes are the reaction products of 4,4'-diphenylmethane diisocyanate and various polypropylene glycol diol/

triol blends. Crosslink density was controlled by varying the ratio of diol to triol. The dynamic mechanical data was fit to the Havriliak-Negami dispersion relation. The most significant effect of increased crosslinking was to increase the low-frequency modulus. A slight increase in relaxation time was observed as crosslink density was increased over the given range, which approximately correlates with the modest change in glass transition temperature seen in this series. Values for the high-frequency modulus and asymmetry of the dispersion were not effected by crosslink density. Comparisons are also made with the sound absorption height and width limits predicted for uncrosslinked polymers [Hartmann *et al.*, *J. Acoust. Soc. Am.* **101**, 2008–2011 (1997)]. [Work supported by the Carderock Division of the Naval Surface Warfare Center's In-house Laboratory Independent Research Program sponsored by the Office of Naval Research.]

5:00

5pPab9. Kinetics of halogenated organic compounds degradation by ultrasound. Hui-Ming Hung, A. J. Colussi, and M. R. Hoffmann (W. M. Keck Labs., California Inst. of Technol., Pasadena, CA 91125)

The kinetics of degradation of CH_2Cl_2 , CHCl_3 , CCl_4 , C_2Cl_4 , C_2Cl_6 , CH_3CCl_3 , and $\text{CHCl}_2\text{CH}_2\text{Cl}$ as induced by ultrasonic irradiation in Ar-saturated solution at frequencies of 205, 358, 618, and 1078 kHz were investigated. Optimal degradation rates were obtained at 618 kHz. At the same frequency, the degradation rate constants for the chlorinated hydrocarbons increases with their Henry's law constants due to the enhanced rectified diffusion. A quantitative analysis of rectified diffusion from Eller-Flynn's model within the framework of recently developed sonochemical observations is able to account for the experimental results [T. G. Leighton, *The Acoustic Bubble* (Academic, New York, 1994), pp. 379–438]. The model predicts a sonochemical reaction efficiency of about 20% for these halogenated hydrocarbons. [Work supported by ONR and Argonne National Laboratory.]

Session 5pPac

Physical Acoustics: Porous Media

James M. Sabatier, Cochair

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Walter Lauriks, Cochair

Department Natuurkunde, Katholieke University Leuven, Physics Department, Celestijnenlaan 200D, B-3001 Leuven, Belgium

Contributed Papers

2:00

5pPac1. Coupled-phase model for the acoustical properties of granular materials. Keith Attenborough and Olga Umnova (Dept. of Mech. Eng., Faculty of Technol., The Open Univ., Milton Keynes, MK7 6AA, UK, O.V.Umnova@open.ac.uk)

The external flow approach has been used to predict acoustical properties of granular materials, assuming the constituent particles are identical and spherical. Analytical expressions for steady-state permeability, high-frequency limit of tortuosity, and characteristic viscous dimension have been derived using the Kuwabara cell model to allow for hydrodynamic interactions between the particles. Good agreement between theoretical and experimental [E. Charlaix, A. P. Kushnick, and J. P. Stokes, *Phys. Rev. Lett.* **61**, 1595–1598 (1988)] results has been demonstrated for stacked glass beads. A new similarity relationship between complex density and compressibility has been derived. It is analogous to that obtained earlier for materials with straight cylindrical pores [M. R. Stinson and Y. Champoux, *J. Acoust. Soc. Am.* **91**, 685–695 (1992)]. In addition to hydrodynamic interactions between particles, thermal interactions have been considered using the “mirror approximation” [Y. Hemar and N. Hermann, *J. Phys. II France* **7**, 637–647 (1997)]. The characteristic impedance of the stacked glass beads layer has been calculated using this relationship and is shown to be in good agreement with measured values and the predictions of pore-based models. [Work supported by EPSRC.]

2:20

5pPac2. Physics of acoustic-to-seismic coupling and detection of buried objects. James M. Sabatier and Ning Xiang (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677, sabatier@olemiss.edu)

Airborne acoustic waves coupled into the surface of the ground excite Biot type I and II compressional and shear waves. If a mine-like target or other inhomogeneity is present below the surface, the ground surface vibrational velocity will show distinct changes due to reflection and scattering of these waves. Sound waves with a wavelength comparable to the object size are suitable for recognizing geometrical shapes of targets, while true wave-like acoustic scattering phenomena can be observed with a shorter acoustic wavelength. In this paper, a review of porous material physics relevant to mine detection will be presented. Recent development in the acoustic technology for mine detection will be reported. Taking advantage of a noncontact measurement technique, the surface vibrational velocity is detected with a laser Doppler vibrometer. [This work is supported by U.S. Army Communications-Electronics Command.]

2:40

5pPac3. Prediction and measurement of the transmission coefficients of transversely isotropic porous plates. Achour Aknine and Bernard Castagnède (I.A.M., Univ. du Maine, Ave. Olivier Messiaen, 72085 Le Mans Cedex 9, France, aknine@laum.univ-lemans.fr)

The prediction of the transmission coefficients is based on the Biot theory of elastic waves propagation in fluid-saturated porous solids. An elastic wave impinging upon an interface between a fluid and a fluid-saturated anisotropic porous solid can generate two compressional waves (one fast and one slow) and two shear waves (one horizontal and one vertical) in the porous medium. The motion in the porous plate results in coupled differential equations involving separate displacements in the skeleton and the saturating fluid. The wave speeds of the propagation modes in the porous medium were determined by seeking the solution of the equations of motion as plane acoustical waves. The transmission coefficients were finally obtained from boundary conditions at the interfaces between the fluid and the porous plate. They were experimentally determined, in the case of a porous corundum ceramic plate immersed in water, by using an ultrasonic immersion technique. The agreement between the predicted and measured transmission coefficients versus the angle of incidence for this plate is acceptable and has been achieved using adjustable and fitted parameters.

3:00

5pPac4. About Biot's theory and the acoustic propagation in aerogels. Luc Forest, Vincent Gibiat (Lab. Ondes et Acoust., URA CNRS 1503, ESPCI, 10 rue Vauquelin, 75231 Paris Cedex 05, France, luc.forest@espci.fr), and Thierry Woignier (Univ. Montpellier II, 34095 Montpellier Cedex 05, France)

Although aerogels are now well known for their very particular acoustical behavior (very low speed of sound, very low acoustical impedance), no theoretical model has been proposed to describe both velocity and absorption. Since aerogels are porous media constituted by an amorphous silica skeletal and air-filled pores, it seems interesting to study the possible application of the current reference theory of propagation in elastic porous media, developed in the 1950s by Biot. Biot has proposed a semiphenomenological theory in which the average motion of both solid and fluid parts is studied separately. Then he introduces in Lagrange's equations the energies of two interpenetrating “effective media,” considering each phase, solid and fluid, as homogeneous. Viscous losses are taken into account from Poiseuille's model. The prediction concerning the velocities in the ultrasonic range are shown accurate and a simple rheological model derived from this theory is defined. Concerning the attenuation, theoretical results and experimental data are carefully compared in the frequency range (0.8–8 MHz). A qualitative agreement for the low-density samples is obtained and it will be shown that the agreement could become quantitative if a dynamic permeability higher than the static one is taken into account.

5pPac5. Influence of static pressure on ultrasound propagation in plastic foams. Christophe Ayrault, Alexei Moussatov, and Bernard Castagnède (LAUM, UMR CNRS 6613, IAM, av. O. Messiaen 72085 Le Mans Cedex 9, France)

The present study deals with the response of an air-saturated porous absorbing material (plastic foams) to external static pressure variation in air. Dynamics of two essential characteristics—acoustical index of refraction and transmission coefficient—are presented both analytically and experimentally. It is shown that the squared index of refraction and the logarithm of transmission coefficient depend linearly on inverse square root of pressure. The standard Johnson–Allard equivalent fluid model for porous media has been utilized. Experimental data in the pressure range from 0.2 to 5 bar at 70 and 84 kHz agree well with the model and thus provide background to a method allowing one to determine experimentally some constitutive parameters of the model. Oncoming research will define the limits of the suggested method. Some of the limits might be due to skeleton vibrations which are seldom observed in air. Promising results are obtained in tests performed with highly damping foams. Increased pressure improves the air-transducer coupling and reduces effective attenuation. It was not possible to characterize such foams earlier by ultrasonics. In addition, it does not affect scattering losses in media, thus providing an opportunity for separating scattering contribution from other losses.

5pPac6. Propagation of an ultrasonic impulse in porous media having a rigid frame. Zine el Abidine Fellah, Achour Aknine, Bernard Castagnède, and Claude Depollier (Lab. Acoust., Univ. du Maine, 72085 Le Mans Cedex 9, France)

The sound propagation in porous materials having a rigid frame filled by air, is well described by the equivalent fluid model where the interactions between fluid and structure are taken into account in the dynamic tortuosity $\alpha(\omega)$ and in the dynamic compressibility $\beta(\omega)$ defined by the basic equations

$$\rho \alpha(\omega) \frac{\partial \langle \mathbf{v} \rangle}{\partial t} = -\nabla \langle p \rangle, \quad \frac{\beta(\omega)}{K_a} \frac{\partial \langle p \rangle}{\partial t} = -\nabla \langle \mathbf{v} \rangle.$$

In the domain of the low-frequency approximation, the behavior of these response factors leads to a wave equation with a dissipative term due to the viscous effects. For high frequencies, a porous material becomes a dispersive medium in which the phase and group velocities are functions of frequency; in such a material a transient pulse changes its shape by spreading. A theoretical model of the sound propagation in a dispersive porous material is presented. This problem is posed in the time domain for an ultrasonic pulse in a slab of porous material. A method is proposed for computing the sound field in the medium. This allows one to deduce the transmission and reflection coefficients which are dependent only on the physical parameters of the medium (but not of the incident wave), which is important for inverse problems. The spreading of the transient pulse is calculated for various porous media and compared to experimental data.

5pPac7. Calorimetric measurements of losses in mineral wool. Jonas Brunsog, Dag Holmberg (Dept. of Eng. Acoust., LTH, Lund Univ., P.O. Box 118, SE-221 00 Lund, Sweden), and Lars Wadsoe (Lund Univ., SE-221 00 Lund, Sweden)

Measurements of the heat generated in mineral wool during a steady-state single-frequency dynamic process are presented. A microcalorimeter is used to register the heat. The calorimeter consists of a heat sink, i.e., a large water container held at constant temperature, a measurement volume, and a temperature gradient sensor in between. The temperature gradient is then related to the heat flux. The sensitivity of the equipment is on the order of 1 microwatt. A small rod is connected to a shaker outside the

calorimeter and to a mineral wool specimen in the other end. The dissipated energy in mineral wool can then be measured during a steady-state single-frequency dynamic process. In order to obtain a value of the reversible energy, one can also measure the acceleration and the voltage to the shaker. Dry friction will cause measurement errors, but if a measurement without any sample is performed, this can partly be compensated for. The heat flux can be related to the dissipation in the mineral wool.

5pPac8. Anisotropy and sound propagation in glass wool. Viggo Tarnow (DTU, AKP, Bygning 358, DK 2800 Lyngby, Denmark)

Sound propagation in glass wool is studied theoretically and experimentally. Theoretical computation of attenuation and phase velocity for plane, harmonic waves will be presented. Glass wool is a highly anisotropic material, and sound waves propagating in different directions in the material will be considered. The computations are based on the geometry of the glass wool that is described by the density of fibers and their diameters. The air drags viscously on the fibers, and movements of the fiber skeleton are important at low frequencies. Propagation of elastic waves in the skeleton is computed by regarding it as a continuous medium described by its elastic moduli and mass density. The computed attenuation of sound waves, for frequencies 50–5000 Hz, will be compared with experimental results for glass wool with fiber diameters of 6.8 micrometers, mass density of 15 and 30 kg/m³, and elastic moduli of 2000 and 16 000 Pa (sound wave vector perpendicular to fibers).

5pPac9. Measurement of ground impedance—Test of method. Hans G. Jonasson and Mikael Oegren (SP Swedish Natl. Testing and Res. Inst., Box 857, SE-501 15 Bors, Sweden, hans.jonasson@sp.se)

Within the frame of a Nordtest project, a method to measure the acoustic impedance of ground has been tested. The Nordtest method has been tested by four different measurement teams, each carrying out four measurements on each of four different test sites. The results show good agreement between the different measurement teams. The measurement uncertainty is less than or equal to one ground class. The proposed Nordtest method, which has much in common with a similar standard under preparation by ANSI, is based on the following: A loudspeaker generating pink noise is used. The measurements are carried out in octave bands. The ground is divided into 12 different classes. For each class the level difference has been calculated theoretically using a specified ground impedance model based on the flow resistivity. The class assigned to a certain ground surface is the one yielding the smallest difference between measured and calculated data. The result reported is the ground class. Different indicators are used to assess the quality of the measurement.

5pPac10. The bulk acoustic properties of a porous medium, derived from a single impedance test. Alan Cummings (School of Eng., Univ. of Hull, Hull HU6 7RX, UK, a.cummings@edm.hull.ac.uk)

In the majority of engineering applications, porous sound absorbing media may be treated as equivalent fluids, with vibration of the solid skeleton being neglected. Most measurement techniques for the bulk acoustic properties of such media (e.g., complex characteristic impedance and propagation constant) involve tests on large samples of material, rather complex series of tests on smaller single samples, or surface impedance tests on two small samples of differing thicknesses. Recently, however, a least-squares method [C. Braccasi and A. Bracciali, Appl. Acoust. 54, 59–70 (1998)] was employed to find values of the structure factor and flow resistivity, in a two-parameter model, from a single surface impedance curve. Though the technique proved successful, the two-parameter model is less accurate over a wide frequency range than the well-known empirical curve fitting method of Delany and Bazley. In the present paper, a least-squares method is described for determining the eight empirical

parameters in the Delany and Bazley method from a single surface impedance curve. Examples of its application are given, and it is demonstrated that it gives a better fit to measured data than the two-parameter model.

5:40

5pAc11. Determination of porous material data via two-port measurements. Mats Abom (ABB Corp. Research, 721 78 Vaerstas, Sweden, mats.abom@seccr.abb.se)

The classical model used to describe porous materials of fibrous type is the so-called equivalent fluid model. For typical fibrous materials this model is valid except for low frequencies where solid frame vibrations become important. Since porous materials are widely used in dissipative silencers, the trend for better and better modeling requires a good knowledge of acoustic data for such materials. This means that a complete characterization is necessary to provide input data for, e.g., FEM or BEM codes. As known from theory, this is possible by specifying the complex wave number (Ka) and complex characteristic impedance (Za) for a propagating plane wave. Of course, for a nonisotropic material, these data must be measured in more than one direction. The standard technique to determine Ka and Za is to use a standing wave tube and measure the complex impedance for two samples of different thickness. In this paper an alternative approach is suggested based on the measurement of the acoustic two-port for a single sample mounted in a duct with plane wave propagation. The paper describes the development of such a two-port test rig and discusses the advantages compared to earlier used techniques.

Contributed Posters

These posters will be on display from Thursday to Friday, 18–19 March. Authors will be at their posters from 10:00 a.m. to 12:00 noon on Thursday, 18 March.

5pAc13. Acoustic determination of water distribution in unsaturated soil. Andreas Blum, Ivo Flammer, and Peter Germann (Geographical Inst., Univ. of Bern, Hallerstr. 12, CH-3012 Bern, Germany, blum@giub.unibe.ch)

Sending acoustic pulses through soil columns, one obtains information on the soil's elasticity characteristics and therefore on its state of humidity. In this laboratory water flow was investigated through $30 \times 30 \times 100\text{-cm}^3$ columns of undisturbed and unsaturated soil. The initial peak of the sound wave arriving at the detector is assumed to be ballistically transmitted through the soil. Its frequency is 10 kHz, thus a spatial resolution of about 6 cm is expected. The present aim is to obtain spatial and temporal information about the soil water distribution by a tomography technique. On this poster acoustic absorption and propagation time measurements during water infiltration are presented. The data are compared with conventional soil humidity measurements by time domain reflectometry.

5pAc14. Surface waves above thin porous layers and periodic structures. Walter Lauriks, Luc Kelders (Lab. voor Akoestiek en Thermische Fysica, Dept. Natuurkunde, Katholieke Univ. Leuven, Celestijnenlaan 200D, B-3001 Leuven, Belgium), and Jean François Allard (Univ. du Maine, UMR CNRS 6613, 72085 Le Mans Cedex 9, France)

Airborne surface waves above thin air-saturated porous layers and periodic structures have been studied in the low-ultrasonic frequency domain. These waves were observed performing two different kinds of experiments. Because surface waves are related to a pole in the reflection coefficient, we adapted a setup based on near-field acoustical holography to the frequency range of interest. In this way, it was possible to measure the reflection coefficient of the porous layer as a function of the angle of incidence. The existence of a surface wave was observed for different layer thickness. Second, the velocity of the inhomogeneous waves has been determined directly by time-of-flight measurements of ultrasonic bursts, by phase velocity measurements with a sine signal, and from in-

6:00

5pAc12. Acoustic signal attenuation, velocity, and filtering by beach sand, with different water content. Hasson M. Tavossi and Bernhard R. Tittmann (Penn State Univ., Dept. of Eng. Sci. and Mech., 227 Hammond Bldg., University Park, PA 16802)

The high-intensity acoustic pulses, transmitted by a piezoelectric transducer of center frequency 40 kHz, are sent through a beach sand medium and received, at up to 35-cm depth, by an identical receiver transducer. The received acoustic signals are then amplified and analyzed on computer. Experimental data are collected at different depth and water content of pore spaces between the grains. Experimental results show that sound arrival velocity decreases from 197 m/s for humid sand, to 120 m/s for partially water-saturated sand with air bubbles, which are always present. Acoustic signal attenuation increased by a factor of ten, from humid to partially water-saturated sand. The peak frequencies in FFT of the transmitted acoustic signal and its speed are dependent on compactness of the medium. The peak spectral amplitude is close to 9.7 kHz for humid sand and 7.5 kHz, for sand partially saturated with water. Experimental data on wave dispersion, phase velocity, and attenuation versus frequency, up to 24 kHz, are obtained. Latest results and tentative theoretical interpretation of experimental data, including porosity and grain size distribution, will be presented.

terference patterns of standing waves. The results are compared to predictions using several models. The reflection coefficient above the reticulated polyurethane foams was predicted by an equivalent fluid model, the frame being motionless at the frequencies considered here. For the calculations of the velocity above rectangular- and triangular-groove gratings, we adapted models developed for electromagnetic waves above these structures to acoustics.

5pAc15. Low-frequency acoustic stimulation of fluid flow in porous media. Peter M. Roberts and Arvind Sharma (Los Alamos Natl. Lab., Group EES-4, MS-D443, Los Alamos, NM 87545, proberts@lanl.gov)

Historical Russian and U.S. research has indicated that acoustic or elastic (seismic) waves transmitted into oil reservoirs at frequencies of 1–500 Hz can increase oil production rates. Results from prior field tests on reservoir formations and laboratory experiments on porous rock samples have been largely inconclusive. The underlying physical mechanisms are still speculative. Comprehensive experimental laboratory and theoretical data on the interactions between acoustic waves and fluid flow in porous media are required before the phenomenon can be exploited reliably for enhanced oil recovery or other applications such as groundwater remediation. A specialized laboratory facility was constructed to characterize effects that low-frequency stress oscillations have on permeability and two-phase fluid flow in cylindrical rock and sand samples. Applied mechanical axial stress, axial and radial strains, permeability changes, fluid production rates, and dynamic elastic moduli are all measured during excitation of the samples. Positive results were observed for enhanced permeability, mobilization of trapped immiscible fluid phases, and increased production rates for sandstone-brine-decane and sand-water-trichloroethylene systems. These results will be presented along with discussions of possible physical mechanisms involved. [Work supported by the U.S. Department of Energy, Office of Fossil Energy.]

5pPac16. Damping modelization of acoustic foams at low frequency. Sahraoui Sohbi and Mariez Emmanuel (Lab. d'Acoust. de l'Univ. du Maine, UMR CNRS 6613, 72085 Le Mans, Cedex 9, France)

The polymeric foams are commonly used as sound absorbing materials. These foams can exhibit at low frequency a resonant vibration of skeleton where the fluid damping is important. The two-phase material dynamic behavior is then described by the Biot equations. This paper examines the elasticity of air-filled PU foams through their microstructure. The foam is modeled by an elastic bars periodic network. It is shown that

the orientation distribution in the space of the bars and their geometry control the macroscopic properties and the anisotropy. The study of the solid-phase deformation was based on a model developed for isotropic foams [W. E. Warren and A. M. Kraynik, *J. Appl. Mech.* **55**, 341–346 (1988)]. An approximative evaluation of the permeability was derived from results obtained for regular arrays of parallel solid cylinders [J. E. Drummond and M. I. Tahir, *Int. Multiphase Flow* **10**, 515–540 (1984)]. The objective of this investigation is to achieve the stress/strain and permeability tensors in the particular case of axisymmetric foams.

FRIDAY AFTERNOON, 19 MARCH 1999

ROOM MA043, 2:00 TO 5:40 P.M.

Session 5pPad

Physical Acoustics and Noise: Outdoor Sound

Robin L. Cleveland, Cochair

Department of Aerospace and Mechanical Engineering, Boston University, Boston, Massachusetts 02215, USA

Keith Attenborough, Cochair

Faculty of Technology, The Open University, Milton Keynes MK7 6AA, UK

Contributed Papers

2:00

5pPad1. The eigenray method for rapidly estimating refraction effects in a linear profile atmosphere. Michael J. White (Appl. Res. Assoc., 5941 S. Middlefield Rd., Littleton, CO 80123)

Eigenray solutions to sound propagation in an atmosphere with a linear sound-speed profile offer a convenient technique to estimate the effects of refraction. Eigenrays represent the trajectories of wave fronts that depart from the source and reach a given receiver. In a linear profile, the rays are arcs of circles, and over flat ground with perfect reflection, the paths can be constructed from a series of hops. Improvements to the eigenray technique are proposed that match the exact solution for pressure and phase in the unbounded linear profile. For reflection, the choice of a correct branch in the expression for divergence is necessary so that adjacent receivers with different numbers of eigenrays report the same pressure. Extension of the technique to lend estimates for some more complex situations is straightforward, although it fails to offer an advantage near caustics and focal regions.

2:20

5pPad2. Atmospheric sound propagation: Application to the case of aircraft landing and take off. Cora Cremezi (ADP Laboratoire, Bat 631, Orly Sud 103, 94396 Orly Aerogare Cedex, France, LaboADP@compuserve.com) and Claude P. Legros (LAUTM, Université Toulouse-Le Mirail, 5 allée Antonio Machado, 31058 Toulouse Cedex, France)

Atmospheric sound propagation has been studied for a long time in the case of source and receiver close to the ground. In the case of an elevated source, studies are scarce and influence of the different propagation effects has not been studied in detail. As transport noises become more and more of a great concern, and among other, aircraft noise around airports, it is necessary to study what phenomena really happen. In aeronautics, this knowledge could initiate new thoughts about noise indices. This study is characterized by a great number of parameters (meteorology, source characteristics, atmosphere) and the main difficulties lie in difficult meteorological condition knowledge and aircraft flight condition uncertainties. A simple ray tracing has shown creation of shadow areas, caustics, and source height influence on propagation, according to meteorological profiles. Then, a more complete propagation model based on a parabolic

equation method has been developed. It takes into account aircraft movements, considering an infinite number of fixed sources to represent the flyover. The model and some preliminary results are presented while an experimentation is going on to validate it better. Once validated, it should allow the estimation of noise level probabilities according to meteorological probabilities.

2:40

5pPad3. Long-range acoustic propagation in the nocturnal boundary layer. Xiao Di and Kenneth E. Gilbert (Appl. Res. Lab. and the Grad. Prog. in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804)

It is a common observation that sound travels to long distances in the nocturnal boundary layer (NBL). For such long-range propagation, the cumulative effects of turbulence can be significant even though NBL turbulence is generally much weaker than typical daytime turbulence. In this paper a 3D version is used of the Green's function parabolic equation (GF-PE) to investigate the effects of turbulence on average levels and cross-range correlation in the NBL at distances of up to 10 km. A comparison is made between 2D and 3D predictions for 50 and 100 Hz. In addition, the cross-range coherence is computed at various ranges for the two frequencies. The implications for beamforming on horizontal arrays in the NBL are discussed. [Work supported by the Defense Advanced Research Projects Agency (DARPA).]

3:00

5pPad4. Calculation of sound reduction by a thick noise barrier in downwind conditions. Jens Forssén (Dept. of Appl. Acoust., Chalmers Univ. of Technol., S-41296, Göteborg, Sweden, jf@ta.chalmers.se)

The downwind propagating sound field from a monopole source is solved by applying a Hankel transform in a stratified medium, according to the method by Rasmussen. The influence of the noise barrier is calculated using the equivalent source method with ring sources of zero order placed inside the fictitious surface of the barrier. At the surface the boundary condition is fulfilled by choosing the correct amplitudes of the equivalent sources. The total field is then calculated as the sum of contributions propagated downwind from both the original source and the equivalent sources.

5pPad5. Acoustic scintillations induced by atmospheric turbulence, as imaged with a large planar vertical microphone array. D. Keith Wilson, Calandra R. Tate (U.S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783), David C. Swanson, and Karl M. Reichard (Penn State Univ., P.O. Box 30, State College, PA 16804)

A planar vertical array of 32 microphones (eight elements in the vertical direction, and four in the horizontal; overall dimensions approximately 6 by 3 m) was constructed on agricultural land near the Pennsylvania State University. Several data sets having 20-min duration were collected, for atmospheric states including windy daytime conditions and still nighttime conditions. The received signals due to a source 770 m distant from the array were processed to form a time-varying virtual image of the source. During the daytime, the source image undergoes dramatic scintillations and fluctuations in its apparent position. The nighttime image is quite stable, and the vertical dependence of the microphone signals suggests that the sound energy propagates as well-defined modes. Cross coherence between the microphones, as a function of vertical and horizontal separation, is also discussed.

3:40–4:00 Break

4:00

5pPad6. Sound propagation near the ground in a turbulent atmosphere. V. E. Ostashev and G. H. Goedecke (Dept. of Phys., Box 30001, New Mexico State Univ., Las Cruces, NM 88003-8001)

For many geometries of sound propagation in an atmosphere, the sound-pressure field is a sum of the direct wave from the source to the receiver and that reflected from the ground. The resulting field is significantly affected by atmospheric turbulence. A theory is presented that gives an analytical formula for the mean-square sound pressure in an anisotropic turbulent atmosphere with temperature and wind velocity fluctuations. This formula contains a “turbulence” factor T which describes the reduction of interference maxima and minima due to atmospheric turbulence. For the important particular case of isotropic turbulence, T is expressed in terms of the coherence function of the direct wave from the source to the receiver. The turbulence factor T and the mean-square sound pressure are calculated and compared for Kolmogorov, Gaussian, and von Karman spectra of temperature and wind velocity fluctuations. Furthermore, the relative contributions to the mean-square sound pressure due to temperature and wind velocity fluctuations are compared. [This material is based upon work supported by the U.S. Army Research Office under Contract No. DAA G55-98-1-0463.]

4:20

5pPad7. The high-power acoustical warning systems. Branko Somek, Sinisa Fajt, and Hrvoje Domitrovic (Faculty of E. E. and Computing, Dept. of Electroacoustics, Unska 3, HR-10000 Zagreb, Croatia, branko.somek@fer.hr)

The high-power acoustical warning and informing systems are necessary for various aims. Mostly these systems work in open space, particularly in urban areas (towns, villages, river valleys). The useful coverage depends upon used transducers-emitters and installed electrical power. Almost all such systems have omnidirectional coverage, i.e., equal acoustical energy is radiated in all directions. The environmental conditions (air temperature and humidity, strength and direction of wind, etc.) and miscellaneous barriers (buildings, land configuration, trees) have an influence on the sound propagation for warning and informing systems. The other important point is that these systems are also used for spoken messages, not only for alarm. The computer simulation was done which takes all these parameters into account, and enables calculation of needed acoustical and electrical parameters.

5pPad8. Infrasound detector design for recording sonic booms and events of CTBT interest. Hein Haak (Royal Netherlands Meteorological Inst., Dept. of Seismology, P.O. Box 201, 3730 AE De Bilt, The Netherlands) and Láslo Evers (Royal Netherlands Meteorological Inst., Dept. of Seismology, P.O. Box 201, 3730 AE De Bilt, The Netherlands)

The Royal Netherlands Meteorological Institute (KNMI) is involved in a project concerning the detection of nuclear explosions and sonic booms. The main area of research and development is concerned with the infrasonic instruments. This includes the pressure sensor, electronics, and the noise-reducers. An optimal array-layout is necessary, next to a suitable instrument, to successfully record infrasonic events. Therefore, the theoretical design of the array and practical implementation at Deelen are of main concern for the acquisitional part of the project. Once data are recorded, the data processing and interpretation has to be done. An on-line data analyses tool was developed to distinguish an infrasonic event from noise and other useless signals. The instruments are based on the Validyne Differential Pressure devices. To create an instrument sensitive in the 0.01–100 Hz frequency range, several constructional efforts were made p.e. acoustic capillary, backing volume, and signal inlet volume. The signal inlet volume is coupled to a noise-reducer, porous hoses or pipes. The symmetrical layout of the hoses/pipes guarantees an improvement in signal-to-noise ratio through spatial integration. The array of 16 microbarographs has an aperture of 2 km. Array techniques, i.e., slowness analyses, were applied to determine the most optimal layout.

5:00

5pPad9. Wind induced infrasound in the atmosphere. Jin Lai Xie (Computer Information and Network Ctr., The Chinese Acad. of Sci., Beijing, PROC 100080) and Zhao Hua Xie (The Chinese Acad. of Sci., Beijing, PROC 100080)

Characteristics of atmosphere infrasound noise field have been studied by measuring and analyzing the infrasound noise field in the atmosphere. A set of space correlation radii of noise corresponding to wind speed 4 m/s has been obtained, which can be taken as the tentative basis for an acoustical array design. At the same time, the wind speed-pressure relation curves have been measured and calculated, and the antijamming ability of the array receiving infrasonic signals and noise-reducing tube is obtained. The following conclusions can be obtained from the present calculations, waveform diagrams, and the field measurements: (1) The spatial correlation radii of the noise caused by the wind are small. (2) The measured curves for wind noise are in accordance with Bernoulli theorem basically. (3) The correlation detection is advantageously available for antinoise. (4) The different averaging duration of the wind noise will cause different results. (5) The main source of natural noise background on ground is the aerodynamic noise produced by local surface wind. The main period lies within 1 to 100 s.

5:20

5pPad10. Snell's law of refraction and sound rays for a moving medium. D. Hohenwarter and F. Jelinek (Inst. of Technol., Dept. for Res. and Testing, Vienna, Austria)

Fermat's principle of least time is used to calculate a new law of refraction for a stratified medium moving horizontally with different temperatures in each layer. This new law of refraction includes velocity of sound, wind speed, and the angle between the vectorial sum of sound velocity and the wind speed. This new equation is compared with the usual approximations for the different refraction laws of a moving medium occasionally mentioned in literature as “Snell's law for a moving media.” The sound rays in a moving thermically stratified medium are refracted more or less (dependent on either upwind or downwind sound propagation), then calculated according to the “Snell's law.” For upward-oriented sound rays and a moderate thermal stratification with high wind speeds,

the difference in the angle of refraction between the new refraction law and the usual approximation is roughly 3 degrees and neglectable for grazing incidence. The sound ray trajectories expressed as analytical equations for a constant sound velocity combined with a wind speed profile

proportional to height; the results are shown in a figure. Furthermore, the sound ray trajectories for a linear sound speed profile combined with a linear height-dependent wind speed profile are calculated and shown graphically.

FRIDAY AFTERNOON, 19 MARCH 1999

ROOM H1028, 2:00 TO 6:20 P.M.

Session 5pPPa

Psychological and Physiological Acoustics: Pitch, Loudness and Frequency Effects

Robert L. Carlyon, Chair

MRC Cognition and Brain Sciences Unit, 15 Chaucer Street, Cambridge CB2 2EF, UK

Contributed Papers

2:00

5pPPa1. Adaptation to a two-component amplitude modulator. Hideki Iwasawa (NTT Adv. Technol., Atsugi, Kanagawa, 243-0198, Japan, iwasawa@av-hp.brl.ntt.co.jp) and Makio Kashino (NTT Basic Res. Labs., Atsugi, Kanagawa, 243-0198, Japan)

Threshold shifts of amplitude modulation (AM) detection caused by adaptation to single- and two-component amplitude modulators were determined. The single-component modulator was sinusoid at either 4, 17, or 21 Hz. The two-component modulator was a complex of sinusoids at 17 and 21 Hz. The carrier was sinusoid at 1000 Hz for both adapting and test tones. Five adults with normal hearing participated in the 6-day experiment. Adaptation to modulators elevated thresholds of AM detection of the same and nearby frequencies relative to the unmodulated-adaptor or no-adapting conditions. In contrast to the findings in modulation detection interference (MDI), where the low-frequency beat of a masker with a two-component modulator interfered with detection of AM at the beat frequency [S. Sheft and W. A. Yost, *J. Acoust. Soc. Am.* **102**, 1106–1112 (1997)], the two-component amplitude modulator did not affect the threshold for detecting AM at the beat frequency. The present results suggest that envelopes are not detected as such, but resolved into components, and that the mechanism responsible for AM adaptation is earlier in processing level than that involved in MDI.

2:20

5pPPa2. A privileged perceptual status for rising intensity tones. John G. Neuhoff (Dept. of Psych., Lafayette College, Easton, PA 18042-1781)

Recent cross-modal matching experiments have shown that listeners reliably overestimate the change of rising level tones relative to equivalent falling level tones [J. G. Neuhoff, *Nature* **395** (6698), 123–124 (1998)]. In a natural listening environment this overestimation could provide a selective advantage, because rising intensity can signal source movement toward an organism. In the present work, listeners heard equivalent rising and falling level tones, and in a three-alternative forced-choice task, indicated which sound demonstrated the greatest change in loudness, or whether the amount of loudness change was the same. If one sound was judged to change more than the other, listeners indicated the magnitude of disparity in loudness change between the two sounds. Results indicate a privileged perceptual status for rising intensity tones, and a greater perceptual disparity between rising and falling intensity tones as overall level increases. These results suggest an asymmetry in the neural coding of tonal rising and falling intensity sweeps, and may be indicative of localization priorities in a natural listening environment.

2:40

5pPPa3. Effects of instantaneous phase shifts and gap duration on pitch of unresolved complex tones. Louise J. White (Lab. of Exp. Psych., Univ. of Sussex, Falmer, Brighton BN1 9QG, UK, louisew@biols.susx.ac.uk) and Christopher J. Plack (Univ. of Essex, Colchester CO4 3SQ, UK)

Introducing a phase change between two short bursts of a sinusoid can produce a pitch shift which corresponds to a shift in the peak of the long-term spectrum [I. V. Nabalek, *Acustica* **82**, 531–539 (1996)]. The present experiment examined this effect with a complex tone consisting of unresolved harmonics where the phase change was applied to the envelope of the complex. The hypothesis that a long-term pitch mechanism sensitive to envelope phase is reset in response to a discontinuity in the stimulus was tested: The stimulus was an unresolved complex tone consisting of the harmonics of a 250-Hz F_0 filtered between 5500 and 7500 Hz. Two 20-ms bursts separated by short gaps were presented; the starting phase of the first burst, the phase between the bursts, and the duration of the gap were varied. A pitch-matching paradigm was used. If the gap between the bursts causes the mechanism to reset, the resetting hypothesis predicts that a phase manipulation of the second burst will not alter the pitch of the complex. Pitch shifts were obtained which do not support a resetting hypothesis, but rather a mechanism in which the mean pulse rate is used to determine pitch.

3:00

5pPPa4. Fundamental frequency discrimination: Influence of lateralized temporal fringes. Hedwig Gockel, Robert P. Carlyon (MRC Cognition and Brain Sci. Unit, 15 Chaucer Rd., Cambridge CB2 2EF, UK), and Christophe Micheyl (UPRESA CNRS 5020, Hopital E. Herriot, 69437 Lyon Cedex 03, France)

Fundamental frequency (F_0) discrimination can be impaired substantially by the presence of another complex (the “fringe”) immediately before and after the target complex. This has been attributed to listeners over-integrating information about the fringe F_0 when estimating the target F_0 . It has been shown [C. Micheyl and R. P. Carlyon, *J. Acoust. Soc. Am.* (in press)] that for the impairment to occur (i) target and fringes have to be in the same frequency region; (ii) if all harmonics of target and fringes are unresolved then they may differ in F_0 ; otherwise, they have to be similar. The present experiments investigated the effect of lateralized fringes. In a 2I-2AFC adaptive procedure, difference limens for F_0 for a 100-ms harmonic target complex were measured in the presence and absence of 200-ms harmonic fringes. The nominal F_0 was 88 or 250 Hz. Stimuli were bandpass-filtered between 125 and 625, 1375 and 1875, or 3900 and 5400 Hz. The target was presented monaurally, while the fringes were monaural (ipsilateral or contralateral) or binaural (diotic or dichotic, lateralized by ILD or ITD). Results showed reduced effects of fringes

localized away from the target, the exact size depending on the resolvability of the components and their specific lateralization. [Work supported by Wellcome Trust.]

3:20

5pPPa5. Competition between space and time in the auditory system.

Robert P. Carlyon (MRC Cognition and Brain Sci. Unit, 15 Chaucer Rd., Cambridge CB2 2EF, England, bob.carlyon@mrc-cbu.cam.ac.uk) and Laurent Demany (Univ. Bordeaux 2, F-33076 Bordeaux Cedex, France)

Listeners were presented with a dichotic 1-s bandpass-filtered (3900–5300 Hz) pulse train at a rate of Fr Hz in one ear and $2Fr$ in the other, against a noise background. Each pulse in the Fr train was simultaneous with a pulse in the other ear. Listeners adjusted the ILD of a bandpass-filtered noise to match the perceived location of one pulse train, and then adjusted the rate of a diotic train, presented at that ILD, to match its perceived rate or pitch. At low rates (e.g., $Fr=1$ Hz) they heard pulses alternating between the midline and the ear receiving the $2Fr$ train, with the perceived rate in each location approximately equal to Fr . At high rates (e.g., $Fr=100$ Hz) they heard a pitch of Fr in one ear and $2Fr$ in the other. At intermediate rates ($12 < Fr < 50$ Hz) a duplex region occurred, whereby the Fr train was heard close to the midline but where the $2Fr$ train was perceived at its “correct” rate. This duplex region, combined with other results, argues against simple schemes in which binaural localisation occurs either entirely “before” or “after” sequential binding processes.

3:40

5pPPa6. Forgetting pitch and loudness. Sylvain Clément, Laurent Demany, and Catherine Semal (Lab. de Neurophysiologie, UMR CNRS 5543, Univ. Bordeaux 2, 146 rue Leo-Saignat, 33076 Bordeaux, France)

Short-term memory for pitch and loudness was investigated in discrimination experiments using a 2I-2AFC paradigm with feedback. The two stimuli presented in each trial were separated by a variable delay (D); they consisted of pure tones, a series of resolved harmonics, or series of unresolved harmonics mixed with lowpass noise. A roving procedure was employed in order to minimize the influence of context coding [N. I. Durlach and L. D. Braida, *J. Acoust. Soc. Am.* **46**, 372–383 (1969)]. During an initial training phase, frequency and intensity discrimination thresholds [$P(C)=0.80$] were measured with an adaptive staircase method while D was fixed at 0.5 s. The corresponding physical differences (in cents or dB) were then constantly presented at four values of D : 0.5, 2, 5, and 10 s. In the case of intensity discrimination, d' markedly decreased when D increased from 0.5 to 2 s, but was not further reduced when D was longer. In the case of frequency discrimination, the decline of d' as a function of D was significantly less abrupt. Similar decline functions were observed for 50- and 500-ms pure tones. These results support the idea that pitch and loudness are processed in separate modules of auditory memory.

4:00–4:20 Break

4:20

5pPPa7. Loudness recalibration and complex tones. Dan Mapes-Riordan and William A. Yost (Parmly Hearing Inst., Loyola Univ. of Chicago, 6525 N. Sheridan Rd., Chicago, IL 60626, dmapes@luc.edu)

Loudness recalibration occurs when a loud (recalibration) tone at frequency f_1 precedes quieter test tones at frequencies f_1 and f_2 . Previous experiments [Mapes-Riordan and Yost, *J. Acoust. Soc. Am.* **101**, 3170(A) (1997)] have shown that the recalibration tone can decrease the loudness of the test tone at f_1 by more than 6 dB. The current experiments addressed loudness recalibration when the test signal and/or recalibration signal was harmonic complex tones. In the first experiment an adaptive tracking procedure measured the equal loudness point between a harmonic complex and a pure tone. In the recalibration conditions, the loudness comparisons were preceded by a recalibration signal consisting of various combinations of frequencies contained in the harmonic complex, or by a pure tone corresponding to the pitch of the missing fundamental. In the

second experiment, listeners were asked to adjust the level of a single harmonic (f_3) in a harmonic complex (f_1-f_5) until it was heard as a separate tone in conditions with and without a recalibration tone at f_3 . The results of these experiments will be discussed in terms of the loci of loudness recalibration relative to perceptual stream formation. [Work supported by a NIDCD Program Project Grant.]

4:40

5pPPa8. Is the auditory filter optimal? Bruce A. Schneider (Univ. of Toronto, Toronto, ON L5L 1C6, Canada)

Gabor [J. I. E. E. London **93**, 429–457 (1946)] showed that a narrow-band filter, with a Gaussian impulse-response envelope, provided the optimal compromise between good temporal resolution and good spectral resolution. The impulse-response function of this optimal Gaussian filter, however, begins before the impulse is applied, and, therefore, is not physically realizable in the auditory domain. A variety of physically realizable impulse response functions were examined to determine which one came closest to the optimal compromise achieved by the Gaussian filter. It is shown that a filter, whose impulse response function is either a gamma function or a generalized Rayleigh function, comes closest to being optimal. Moreover, these two filter types provide good fits to the data from notched-noise experiments.

5:00

5pPPa9. Adaptive psychophysics based on unforced-choice tasks.

Christian Kaernbach (Inst. für Allgemeine Psych., Univ. Leipzig, Seeburgstr. 14-20, 04 103 Leipzig, Germany)

Perceptual thresholds are usually measured by means of adaptive staircase procedures. The latter are commonly combined with forced-choice tasks in order to avoid problems arising from unstable criteria in yes/no tasks. The present study suggests the use of “unforced-choice” tasks, when the observer is given an additional response alternative, “don’t know.” Simulations and experimental data are presented which demonstrate that if combined with an appropriate adaptive rule, the unforced-choice paradigm can be at least as efficient as forced-choice tasks, if not superior.

5:20

5pPPa10. Correlations between speech processing and auditory functions at 1 kHz.

Ingrid M. Noordhoek, Tammo Houtgast, and Joost M. Festen (Dept. of Otolaryngol., Univ. Hospital VU, P.O. Box 7057, 1007 MB Amsterdam, The Netherlands, im.noordhoek@azvu.nl)

Even when all relevant speech information is presented above threshold, intelligibility may still be reduced for hearing-impaired listeners. An attempt is made to relate impaired speech processing to a deterioration of specific auditory functions. Performance of hearing-impaired listeners is measured on psychoacoustic tests concerning temporal and spectral resolution, frequency and intensity discrimination, and temporal and spectral integration. All auditory functions are measured at 1 kHz. Speech reception is measured with the SRBT test (speech reception *bandwidth* threshold). This test determines the minimum width of a speech band with a center frequency of 1 kHz required for 50% intelligibility. A wider-than-normal SRBT can simply be the consequence of inaudibility of part of the speech signal. To account for this possibility, each SRBT is converted to an SII (speech intelligibility index, the new articulation index). This SII may be interpreted as the proportion of the total speech information required by the listener for 50% intelligibility. An elevated SII is considered an indication of impaired processing of suprathreshold speech. This may be caused by the deterioration of specific auditory functions. Relations between the performance on psychoacoustic tests and the (elevated) SII will be examined. [Work supported by the Foundation “Heinsius-Houbolt Fonds.”]

5pPPa11. Further studies of auditory decreruitment and a visual analog. Georges Canévet, Xavier Regal, Olivier Sauvage (CNRS-LMA, 13402 Marseille Cedex 20, France), Robert Teghtsoonian, and Martha Teghtsoonian (Smith College, Northampton, MA)

A tone continuously decreasing in amplitude from moderate to weak levels will, at any point in the sweep, have a loudness increasingly lower than an equally intense tone presented alone. This accelerated loss in loudness has been called decreruitment [Canévet and Scharf, *J. Acoust. Soc. Am.* **88**, 2136 (1990)] and interpreted in part as an adaptation effect. For a 1-kHz tone, the amount of decreruitment increases with the duration of the sweep and becomes asymptotic at about 20 s; a phenomenon similar to decreruitment can be observed for circles continuously shrinking in size [Teghtsoonian *et al.*, P & P, 1998, submitted]. Further studies, in which tone frequency and sensation level are varied, show that (a) at 4 kHz, the degree of decreruitment increases with sweep duration for a longer period (over 50 s) and achieves a higher value than obtained with a 1-kHz tone, and (b) sensation level has little effect on either measure. These findings for loudness are not easily interpreted in terms of simple adaptation but may be consistent with a form of adaptation characterized as "medium-term." A study confirming the accelerated shrinkage of a continuously decreasing circle is also reported and the problem this creates for adaptation-based interpretations of loudness decreruitment is noted.

5pPPa12. Effect of time distribution of energy on loudness evaluation. Sabine Meunier (Lab. de Mécanique et d'Acoustique-CNRS, 13402 Marseille Cedex 20, France), Patrick Susini (Inst. de Recherche et Coordination Acoustique/Musique, Marseille, France), and Xavier Regal (Lab. de Mécanique et d'Acoustique-CNRS, 13402 Marseille Cedex 20, France)

Most work on loudness have been done on stationary sounds. However, environmental sounds are usually temporally variable, and memory effects could be important in loudness evaluation of long-lasting dynamic sounds. The loudness of one-minute sounds containing a local dominating peak of energy was measured. The sounds differed only by the temporal position of the peak. The influence of peak location on the global loudness judgment was evaluated. The global loudness was first evaluated using magnitude estimation. Contrary to expectations, mean estimations were the same for all sounds regardless of peak distribution. In a second experiment, subjects had to judge the loudness, using cross-modal matching. They matched the size of a circle on a computer screen with the loudness of the sound. At the end of the signal, a global estimation was also made using the same cross-modal matching paradigm. This global estimation was made either just after the sound or after a pause of one minute. During the pause, the subject had to perform a distracting task. Relations between global and continuous judgments are discussed. The study can help to design a model of the loudness of temporally variable sounds.

FRIDAY AFTERNOON, 19 MARCH 1999

POSTER GALLERY, 2:00 TO 4:00 P.M.

Session 5pPPb

Psychological and Physiological Acoustics: Binaural Processing, Cocktail Party Effects and Localization (Poster Session)

Kourosh Saberi, Chair

Beckman Laboratory, Caltech, Pasadena, California 91125, USA

Contributed Papers

Posters will be on display from Thursday to Friday, 18–19 March. Authors will be at their posters from 2:00 p.m. to 4:00 p.m. on Friday, 19 March.

5pPPb1. Spectral integration in binaural processing. Kourosh Saberi, Yoshifumi Takahashi, and Haleh Farahbod (Div. of Biol., Caltech, Pasadena, CA 91125)

Results are reported from human psychophysical and animal neurobehavioral experiments that examined the ability of the binaural system to integrate information across frequency channels. The psychophysical experiments measured the lateralization of low-frequency (<1200 Hz) multitone complexes as a function of number of components (five or fewer) and frequency spacing between these components (25 to 400 Hz). A pointing task was used to match the perceived lateral position of a broadband noise burst to that of the complex. The complex contained a fixed interaural delay of 1500 μ s leading to the left, and the pointer's perceived lateral position was adjusted by varying its interaural level difference. Parallel neurophysiological experiments were conducted in the optic tectum (superior colliculus) of the barn owl, using narrow-band dichotic noise bursts and multitone stimuli; the localization behavior of the owl in response to these same stimuli was also measured. Results show complex patterns of responses with implications for models of frequency interac-

tion in the binaural system. Linear multiplicative or additive models customarily used in cross-correlation analysis do not explain the entire data set. [Work supported by NIH.]

5pPPb2. A cocktail party effect in the median plane? Robert S. Bolia (Veridian, 5200 Springfield St., Dayton, OH 45431, rbolia@falcon.al.wpafb.af.mil), Mark A. Ericson, W. Todd Nelson, Richard L. McKinley (Air Force Res. Lab., Wright-Patterson AFB, OH 45433-7901), and Brian D. Simpson (Wright State Univ., Dayton, OH 45435)

Recent research has been conducted on the effects of spatialized audio on a listener's ability to detect and identify a target speech signal when presented among nontarget speech signals in the horizontal plane [W. T. Nelson *et al.*, Proceedings of the 1998 IMAGE Conference (1998), pp. 159–166]. However, the existence of a "cocktail party effect" in the median plane has not been addressed. The purpose of the present investigation was to determine whether or not the spatial separation of multiple simultaneous speech sources in the median plane leads to improved detec-

tion and intelligibility. Independent variables include the number and angular separation of the speech signals, the sex of the target talker, and the presence or absence of head motion cues. All speech signals—phrases from a modified Coordinate Response Measure [T. J. Moore, AGARD Conference Proceedings 311: Aural Communication in Aviation (1981), pp. 2.1–2.6]—were digitally filtered via nonindividualized HRTFs and presented over headphones. Results will be compared with those obtained in the horizontal plane. Implications for the design of auditory displays will be discussed.

5pPPb3. Perceptual learning in the discrimination of interaural time differences. Takayuki Kawashima (Dept. of Psych., Div. of Humanities and Sociology, Univ. of Tokyo, 7-3-1, Hongo, Bunkyo-ku, Tokyo, 113-0033 Japan, lh76038@hongo.ecc.u-tokyo.ac.jp), Makio Kashino (NTT Basic Res. Labs., Atsugi, Kanagawa, 243-01 Japan), and Takao Sato (Univ. of Tokyo, Bunkyo-ku, Tokyo, 113-0033 Japan)

Experiments were conducted to clarify whether spatial resolution improves with practice. Four adult subjects were involved in measuring a discrimination threshold in interaural time difference (ITD) discrimination tasks. The threshold was measured using adaptive, 2-alternative, forced-choice procedure. The subjects were pretested with 500-Hz pure tones which have ± 300 (plus meaning that the right channel signal leads) or 0 μ s ITDs. Then two subjects were repeatedly tested for 9 days (about 80 min per day) at 300 μ s ITD and the others at -300 μ s ITD, with feedback to their responses. Following the practice period they were tested at ± 300 and 0 μ s ITDs again (post-test). At the trained ITDs, the spatial resolution improved throughout practice period, and the threshold declined from 81 to 39 μ s on average between pretest and post-test. At untrained ITDs, on the other hand, the resolution did not improve between two tests. The improvement was maintained even 74 days after the post-test. These results suggest that the spatial resolution based on ITD improves with practice, and the improvement occurs locally in a certain ITD region, at least within 300 μ s around the trained ITD.

5pPPb4. Measuring the role of masker-correlation uncertainty in binaural masking experiments. Armin Kohlrausch and Jeroen Breebaart (IPO-Ctr. for Res. on User-System Interaction, P.O. Box 513, NL-5600 MB Eindhoven, The Netherlands)

Masked thresholds of a 500-Hz sinusoid were measured in an $NpS\pi$ condition for both running and frozen-noise maskers using a 3IFC procedure. The nominal masker correlation varied between 0.64 and 1, and the bandwidth of the masker was either 10, 100, or 1000 Hz. The running-noise thresholds were expected to be higher than the frozen-noise thresholds because of interaural correlation uncertainty of the masker intervals for running-noise conditions. Since this correlation uncertainty decreases with increasing masker bandwidth, differences between running- and frozen-noise conditions should decrease with increasing bandwidth for interaural correlations smaller than +1. For an interaural correlation close to +1, no difference between frozen-noise and running-noise thresholds is expected for all values of the masker bandwidth. These expectations are supported by the experimental data. For the 10-Hz running-noise condition, the thresholds can be accounted for in terms of interaural correlation uncertainty. For the frozen-noise conditions and both the 100- and 1000-Hz running-noise conditions, it is likely that the limits of accuracy of the internal representation determine the detection thresholds. [Work supported by the Dutch Organisation for Scientific Research (NWO).]

5pPPb5. Modeling the role of masker-correlation uncertainty in binaural masking experiments. Jeroen Breebaart and Armin Kohlrausch (IPO-Ctr. for Res. on User-System Interaction, P.O. Box 513, NL-5600 MB Eindhoven, The Netherlands)

Recently, a new approach to modeling binaural interaction was described [Breebaart *et al.*, J. Acoust. Soc. Am. **103**, 2844–2845 (1998)]. This model is based on a subtractive mechanism and is sensitive to interaural time differences as well as interaural intensity differences. In order

to extend this model to masking data for partially correlated noise maskers, the following experimental findings have to be considered. (a) For broadband maskers with correlations < 0.95 , the power ratio between signal and masker difference intensity (i.e., masker intensity at the output of the subtractive mechanism) is constant. (b) In broadband conditions, frozen- and running-noise maskers lead to the same thresholds, indicating that masker correlation uncertainty does not influence detectability of the signal. (c) For narrow-band running-noise maskers, the masker correlation uncertainty is the dominant factor to explain signal thresholds. Result (a) can be implemented by applying a logarithmic compression to the output of the subtractive mechanism and adding an internal noise with a constant rms value. With this modification, the model is capable of describing the differences between frozen- and running-noise maskers as listed under (b) and (c). [Work supported by the Dutch Organisation for Scientific Research (NWO).]

5pPPb6. A priori knowledge of the sound source spectrum in median plane localization. Kazuhiro Iida (AVC Res. Lab., Matsushita Comm. Ind., 600 Saedo, Tsuzuki, Yokohama, 224-8539 Japan, kiida@adl.mci.mei.co.jp) and Masayuki Morimoto (Kobe Univ., Nada, Kobe, 657-8501 Japan)

The previous papers [e.g., J. Hebrank and D. Wright, J. Acoust. Soc. Am. **56**, 935–938 (1974); F. L. Wightman and D. C. Kistler, J. Acoust. Soc. Am. **101**, 1050–1063 (1997)] indicated that the median plane localization mechanism requires *a priori* knowledge of the source spectrum. Listeners, however, localize a sound image accurately for various sound sources on the median plane, such as music, voice, and noise. Since these sources differ from each other in spectrum, it is inferred that listeners have acquired the knowledge of the source spectrum for each sound source. This paper clarifies whether *a priori* knowledge of the sound source spectrum is classified by the kinds of source or not in the median plane localization mechanism.

5pPPb7. Analysis of different pointing methods in localization experiments of sound sources. Christoph Pörschmann and Thomas Djelani (Inst. of Commun. Acoust., Ruhr-Universität Bochum, 44780 Bochum, Germany, poersch@ika.ruhr-uni-bochum.de)

Sound localization experiments are described which aim to analyze and compare different pointing methods for sound-source localization in virtual and real environments. In the first experiment, subjects indicated the perceived direction of sound incidence by pointing directly to the source with their hands. In the second experiment, the direction was indicated using the GELP (Gods eye localization pointing) technique, i.e., by pointing at a spherical model of the auditory space. In the third experiment, a head-pointing task was applied, that is to say the direction of localization was indicated by turning the head toward the direction of the source. All the localization tests were carried out in a virtual auditory environment. Individually measured HRTFs were used for the auralization; the stimuli (pulsed white noise) were presented via headphones. The subjects were allowed to perform small head movements in order to improve the localization capability. The results of the experiments are discussed. The systematic errors that occur when using the different pointing methods are analyzed. It is shown that the localization blur averaged over all the directions is smaller using the GELP technique, while the blur for the front directions is smaller when the subjects point directly to the source.

5pPPb8. Localization experiments with saccadic responses in virtual auditory environments. Heike Heuermann-Mehmood and Hans Colonius (Inst. fuer Kognitionsforschung, FB 5-A6, C. v. Ossietzky, Univ. Oldenburg, 26111 Oldenburg, Germany)

In this study, subjects had to localize sound sources in free-field listening conditions and in virtual environments. Under virtual listening conditions, three auditory displays with distinct degrees of individualization were used. In contrast to previous studies using verbal reports or pointing

techniques, the participants indicated the perceived location of a sound source by turning their eyes toward it. Localization performance under virtual conditions was comparable to free-field conditions for all subjects, at least if individual HRTFs were used. In general, judgments turned out to

be more accurate than in previous studies, especially with regard to dispersion of single judgments. This may be due to a close link between the auditory and the oculomotor system, leading to a quasiautomatic saccade toward an auditory stimulus, even without a corresponding visual target.

FRIDAY AFTERNOON, 19 MARCH 1999

ROOM H2036, 2:00 TO 6:20 P.M.

Session 5pSAa

Structural Acoustics and Vibration: Structural Vibration, Radiation and Scattering II

Donald B. Bliss, Cochair

Department of Mechanical Engineering, Duke University, Hudson Hall Science Drive, Durham, North Carolina 27708, USA

Victor T. Grinchenko, Cochair

Institute of Hydromechanics, Hydrodynamic Acoustics, Zhelyabov Str. 8/4, 252057 Kiev, Ukraine

Contributed Papers

2:00

5pSAa1. The modulation direct field radius: Model. Sven Lindblad and Karl-Ola Lundberg (Dept. of Eng. Acoust., LTH Lund Univ., P.O. Box 118, SE-22100 Lund, Sweden, sven.lindblad@kstr.lth.se)

When experimentally estimating reverberant field parameters, it is important to be positioned outside the "direct field." In an often used model the total energy density is considered to be a sum of two terms, associated with the direct field and the reverberant field, respectively. The first term is diminishing with increasing source-receiver distance, while the second is receiver position independent. At a certain distance the magnitudes of the two terms are equal. In an extended model which allows fluctuating input power, the modulation direct field radius is monotonically increasing with the modulation frequency. As the modulation frequency goes to zero, the radius diminishes to the usual steady-state direct field radius. The reason is that the reverberant field smears out the modulation, especially above a certain modulation frequency, while the modulation in the direct field is maintained. Damping is often estimated by measuring decay times when a constant input power is suddenly switched off. The modulating signal is then one unit step. The magnitude spectrum of such a signal, though falling, has components at all frequencies. The implication of this is that no sufficiently large measuring source-receiver distance exists, using this method.

2:20

5pSAa2. Eigenforms and eigenfrequency spectrum of finite elastic cylinder. Victor T. Grinchenko (Inst. of Hydromechanics, 8/4 Zhelyabova St., 252057 Kiev, Ukraine, vgr@ihm.kiev.ua)

The reflection of elastic waves from free boundaries is accompanied by the effects of transformation of longitudinal waves to transverse ones and vice versa. The transformation is the physical basis for interesting and practically important peculiarities in eigenfrequency spectrum and characteristics of eigenforms. The cases of finite cylinder ($R < L$) and circular plate ($R > L$) are considered to illustrate these peculiarities. The main idea of the boundary problem solution method is described. The method provides a way to get the eigenmode characteristics accurately in a wide-frequency range. The general conclusion of the study is that it is not possible to give a qualitative explanation of the eigenfrequency spectrum and eigenform properties in the scope of the concept of standing waves with respect to propagating ones in long elastic cylinder and infinite layer. The special propagating evanescent waves are important in the eigenmode forming process. The influence of this kind of wave results, for example, in occurrence of eigenfrequencies value, which increases when dimen-

sions of the cylinder increase. Specific features of corresponding eigenforms are discussed. Comparison of the numerical and experimental data is presented.

2:40

5pSAa3. Acoustic pressure radiated by a vibrating body in the high-frequency domain. Eric Landel, Patrick Blanc (Principia R. D., Zone Portuaire de Brgaillon, F-83507 LA Seyne/Mer, France), and Thierry Loyau (INRS, F-54501 Vandoeuvre, France)

To develop methods of predicting radiated noise that can be used in design offices, it is essential to formulate hypotheses to simplify the general problem. The expression of the acoustic pressure autospectrum for quasiplane surfaces in far field is used, obtained from the Green's function, which depends on: the Green's function, the cross-spectral density function of the normal acceleration on the body. The spatial distribution of the vibrational field on the vibrating body is, in an industrial context, very difficult to obtain, as it would require too many measurement points. It is therefore necessary to approximate it, and to this end it is assumed that: the vibrating body can be approximated by a combination of elementary surfaces; these surfaces are thin plates; the energy of each mode is uniformly distributed in a given frequency band. Once the vibrational field has been characterized in this way, it is possible to deduce the acoustic radiated pressure. A number of validation tests were carried out, which demonstrated close agreement between the computed and experimental data in the two following cases: a baffle plate and a box.

3:00

5pSAa4. Finite panel sound radiation through an absorbent layer. Ennes Sarraj (TU Dresden, Inst. fuer Technische Akustik, 01062 Dresden, Germany, sarraj@eakis1.et.tu-dresden.de)

It is often required to predict the sound radiated from a vibration through an absorbent layer (trim). Statistical energy analysis may be applied in such a case. The required radiation coupling loss factor is estimated from the radiation efficiency. To get the radiation efficiency for the trimmed panel, that of the panel (into air, without trim) is combined with the transmission loss of the absorbent layer to give a corrected radiation efficiency. Although this procedure is quite straightforward, it may produce poor results. An alternative and more accurate procedure for the estimation of the radiation efficiency is presented. The absorbent layer is fully considered in the calculation, which is done using a wave number integration technique. The results from both methods show remarkable differences.

5pSAa5. Low-frequency ($ka < 10$) monostatic scattering and elastic wave generation produced by end insonification of homogeneous, finite rods ($l/a = 8$). Timothy J. Yoder (SFA, Inc., Largo, MD), Joseph A. Bucaro, and Brian H. Houston (Naval Res. Lab., Washington, DC)

A parametric study of the low-frequency ($ka < 10$) monostatic scattering and elastic wave generation produced by end insonification of homogeneous, finite rods ($l/a = 8$) as a function of their material parameters (density, Young's modulus, and Poisson's ratio) is presented. The monostatic scattering and surface displacement are calculated via finite/boundary elements (verified through experiments performed by Laboratory for Structural Acoustics at NRL) as a function of the rod's material parameters. These parameters are slowly varied to show the acoustic transitions that occur between rods composed of different materials. One backscattering feature is of particular interest because its amplitude is large and it only occurs for certain materials. The appearance (and disappearance) of this highlight can be used for material identification and is explained by the behavior of the fluid-loaded longitudinal wave dispersion characteristics of the rod. The dispersion characteristics are calculated [P. B. Nagy, *J. Acoust. Soc. Am.* **98**, 454–457 (1995)] and compared to the dispersion found by transforming the surface displacement calculated via finite/boundary elements. This comparison identifies the elastic waves generated on the fluid-loaded rod and is ultimately used to identify the cause of the large backscattering feature.

5pSAa6. The modulation direct field radius: Experiments. Karl-Ola Lundberg and Sven Lindblad (Eng. Acoust., LTH, Lund Univ., P.O. Box 118, SE-221 00 Lund Sweden, karl-ola.lundberg@kstr.lth.se)

The modulation transfer function, CMTF, of a linear system can be derived by post processing the impulse-response by band-pass filtering and squaring followed by a Fourier transform. Experiments were carried out in a reverberation chamber of $5.6 \times 6.4 \times 6.1$ m. The source was a small loudspeaker with an approximately omnidirectional radiation. The impulse-responses were measured on axis by means of a microphone along a straight line at 16 different source-receiver distances in the range 0.4–4 m. The corresponding free-field impulse-response was measured in a nonechoic chamber, also on axis at the distance 1.00 m. The postprocessing was performed computationally. The direct field contribution in the reverberation chamber impulse-responses was excluded by subtracting the response by the adjusted free-field impulse-response. The adjustment consisted of amplitude scaling and left or right shifting, depending on the actual source–receiver distance. The CMTFs were computed for the adjusted impulse-response and for the adjusted free-field impulse-response. For a selected modulation frequency, the magnitudes of the two derived CMTFs were equal at a certain source–receiver distance, here named modulation direct field radius. The results agree very well with the proposed model.

5pSAa7. Broadband 3-D high spatial density structural response measurements of a pair of framed air-loaded cylinders of differing internal complexity. Joseph F. Vignola (Naval Res. Lab., Washington, DC 20375-5000 and SFA, Landover, MD 20785), Douglas M. Photiadis, and Brian H. Houston (Naval Res. Lab., Washington, DC 20375-5000)

The presence of internal structure can greatly alter the acoustic behavior of elastic structures. The effect of a large number of internal oscillators on the vibrational response of an air-loaded framed cylindrical shell has been examined experimentally. This examination included the construction of two shells of the same design, one of which has 880 attached oscillators. Spatially dense (200 axial points by 80 azimuthal = 16 000 points) 3-D laser Doppler vibrometer measurements have been collected for both of these structures for the case of broadband radial point excitation. Comparisons made between these two visually identical structures

reveal dramatic differences. These include a modification of the participation factors between the three displacement polarizations and strong spatial confinement of the vibration response due to complexity. These results are presented together with a discussion of mechanisms which may account for these observations. [Work sponsored by ONR.]

5pSAa8. Measurements of the condition of compliant coatings for prediction of acoustic performance. John W. Doane (GTRI/SEAL, Georgia Tech, M/S 0405, Atlanta, GA 30332, john.doane@gtri.gatech.edu) and Jacek Jarzynski (Georgia Tech, Atlanta, GA 30332)

The goal of this work was to develop a system and analysis procedure for measuring the material properties of a viscoelastic polymer slab. This is accomplished by exciting a transient acoustic wave in the slab which has been bonded to a steel plate and observing the wave's propagation along the surface of the slab. This propagation is modeled numerically and an iterative analysis process yields the properties of interest. The system is a scanning laser Doppler vibrometer which is capable of measuring the particle velocity on the surface of a reflective target. The scanning feature allows for data to be taken rapidly at a number of locations via a computer-driven algorithm which also digitizes and stores the data for post-processing analysis. There are two numerical models developed for this problem, one based on geometrical ray acoustics and the other based on inverse Fourier and Hankel transform methods. The ray model is developed to give a quicker and relatively simpler way of investigating the physics of the problem whilst the inverse transform method gives a more robust and detailed method for analyzing the peculiarities of the problem.

5pSAa9. Analysis of structure-borne sound transmission using spatial statistical distributions. Ross A. Fulford (Inst. fuer Technische Akustik, Einsteinufer 25, 10587 Berlin, Germany, fulf0637@mach.ut.tu-berlin.de)

For a source of vibration, i.e., a machine, connected to a receiving structure, i.e., a floor, the vibrational power transmitted is dependent upon the dynamic characteristics of both the source and receiver. If the source and receiver are connected at N points and with M components of motion, the exact character of each structure can only be quantified using $(MN)^2$ frequency-dependent terms. Most often this leads to an analysis involving a vast amount of data. Upon recognition of such, together with an understanding that the terms have inherent variations, an approximate analysis method is suggested in which the precise nature of the structure's spatial response is disregarded and instead considered as a probabilistic distribution. With the approach, not only is the amount of data required significantly reduced but, moreover, the precision with which it has to be procured is also relieved. Application is demonstrated alongside work in progress with respect to practical implementation.

5pSAa10. The influence of sound radiation on vibration of mechanical structures. Andrzej B. Dobrucki and Bronislaw Zoltogorski (Wroclaw Univ. of Technol., Inst. of Telecommunications and Acoust., Wybrzeze Wyspianskiego 27, 50-370 Wroclaw, Poland)

Two methods of inclusion of coupling between the vibrating structure and produced acoustic field are discussed. The first iterative method consists of calculation of successive corrections of exciting nodal forces caused by sound pressure and solving the FEM equation of the structure. It has been proved that near the resonances of the structure this method can be nonconvergent. The second method consists of calculation of the generalized complex stiffness matrix of the sound field on the surface of the structure and including it into the stiffness matrix of the structure. The resulting stiffness matrix is a full matrix, and the algorithm for solving the FEM equation system is time consuming. It has been shown that near resonances of the structure, the global mechanical power provided to the radiating structure decreases in comparison to the power provided to the

nonradiating structure. Near the antiresonant frequencies, the power provided to the structure with included radiation is greater. On this basis, a simple method of calculation of mechanoacoustical efficiency near the resonances and antiresonances of the structure is derived. The considerations are useful for calculation of efficiency of electroacoustical transducers. The results of calculation for vibrating and radiating shells of revolution are given.

5:40

5pSAa11. Acoustic emission model for a thin circular plate with large deflections. Nicolae Enescu, Mihai Bugaru, and Mihai V. Predoi (Catedra de Mecanica, Universitatea Politehnica Bucuresti, Splaiul Independentei 313, Bucharest, Romania)

This paper presents a model for the acoustic emission of a circular plate, in the domain of large deflections. The thin circular plate is considered homogeneous, isotropic, and fixed on the boundaries. In the hypothesis of large displacements, the nonlinear von-Karman dynamic model was adopted. The external action is periodic and axisymmetric. An approximate analytical method was developed, based on the Kantorovich method and the asymptotic method, to compute the dynamic response of the plate. Starting from the characteristic parameters of the nonlinear vi-

brations of the plate, a model of acoustic emitter based on the Rayleigh formulation was developed, to obtain the acoustical pressure and the spatial directivity characteristic. Experimental tests were developed in an anechoic chamber, in order to confirm the model. The physical model of the plate consists of a circular frame having a high rigidity and a brass plate of 0.2-mm width. In the transverse direction, passing through the mass center of the plate, it acts as an excitatory device and the acoustic emission of the plate is analyzed. In this way, the natural frequencies of the plate and the spatial directivity characteristic have been determined and a satisfactory agreement with the theoretical results has been found.

6:00

5pSAa12. The acoustic power radiated by a circular plate hanged articulately. W. J. Rdzanek (Inst. of Phys., Pedagogical Univ. of Rzeszów, Rejtana 16a, 35-311 Rzeszów, Poland)

Radiation of the acoustic wave by vibrating planar circular plate, hanged articulately into a planar rigid infinite baffle is considered. Axially symmetric, harmonic in time vibrations are investigated. Frequency characteristics of radiated real power are determined. There is achieved the comparative analysis of the real power in the case of articulated hanging of the plate and radiated power by a plate with a rigidly fixed edge.

FRIDAY AFTERNOON, 19 MARCH 1999

ROOM EB301, 2:00 TO 4:20 P.M.

Session 5pSAb International Workshop on Active Noise and Vibration Control

Structural Acoustics and Vibration: Transducers for Active Noise and Vibration Control

Geoff Leventhall, Cochair

Consultant in Noise, Vibration and Acoustics, 150 Cradocks Avenue, Ashtead, Surrey KT21 1NL, UK

L. Eric Cross, Cochair

Electrical Engineering, Pennsylvania State University, 187 Materials Research Laboratory, University Park, Pennsylvania 16802-4800, USA

Invited Papers

2:00

5pSAb1. Novel electrostrictive poly(vinylidene fluoride-trifluoroethylene) copolymer actuators. Q. M. Zhang, Vivek Bharti, Z-Y. Cheng (Mater. Res. Lab., Penn State Univ., University Park, PA 16802), T. Romotowski, Frank Tito (Naval Undersea Warfare Ctr., Newport, RI 02841), and R. Y. Ting (Univ. of Central Florida, Orlando, FL 32816)

Electroactive materials for applications in active noise and vibration control should possess a high displacement capability, high reliability, and be easily fabricated to conform to complicated surfaces and shapes. Apparently, almost all the current available materials cannot meet these requirements. It was discovered recently that a massive electrostriction (strain level ~5%) can be achieved in specially treated P(VDF-TrFE) copolymers [Q. M. Zhang, Vivek Bharti, and X. Zhao, *Science* **280**, 2101 (1998)]. More significantly, because of the high elastic modulus, the material also exhibits high elastic energy and power density, an order of magnitude higher than those from the conventional piezoceramic and magnetostrictive materials. In addition to the large longitudinal strain, the transverse strain can be tuned over a broad range by varying the processing and irradiation conditions, from near zero in unstretched samples to more than the longitudinal strain in stretched samples. In this paper, the electromechanical properties of this class of material and the performance under different external conditions will be presented. The results on the unimorph and multilayer actuator performance fabricated using the new electrostrictive polymers with different electrode materials will also be presented. [Work supported by ONR.]

2:20

5pSAb2. Cymbal-based actuator panels. Thomas R. Howarth and James F. Tressler (Naval Res. Lab., Washington, DC 20375-5350, howarth@nrl.navy.mil)

The cymbal flextensional element has been configured into a "1-3" piezocomposite-type arrangement between two stiff plates to produce a thin packaged actuator device for active noise and vibration control. This presentation will discuss the activities at the Naval Research Laboratory in developing the cymbal-based actuators from the basic element design to useful device applications. This paper

will discuss the effects of spacing, element geometries, and material selections with respect to operating frequency bands and displacement profiles. Results of measured in-air immittance and laser Doppler vibrometry LDV) will be presented to discuss the output profiles. Results of a program for integrating the panels into high force applications will be described and shown. [This research was supported by the Office of Naval Research Code 321SS.]

Contributed Papers

2:40

5pSab3. Estimate of the radiated acoustical power using strain sensing. Helmut Berger (Inst. for Telecommun. and Electroacoust., Darmstadt Univ. of Technol., Merckstrasse 25, 64283 Darmstadt, Germany)

In common active structural acoustic control systems, the acoustical feedback signal is obtained from error microphones placed in the far field of the vibrating structure. Using strain sensors on the surface, the need for far-field acoustic sensors is eliminated, which is advantageous in practical applications where no microphones can be used. An estimate of the radiated acoustic power from a vibrating plate by the use of on-plate PVDF sensors is discussed. Since the number and the location of the sensors directly affect the quality of the radiation information, the influence of various sensor arrangements is investigated. For this purpose, a clamped rectangular plate is considered which is excited at four discrete frequencies in such a way that four modes contribute effectively to sound radiation. Since the vibrational behavior is well known, the required number of PVDF sensors may be reduced, which speeds up the computation of the radiated acoustic power during the adaptation process. It will be shown that the analytical results agree well with experimental data.

3:00

5pSab4. A new piezoelectric active noise reduction earplug. Raoul Bauer, Pascal J. F. Hamery, Pierrick Lotton,^{a)} and Armand L. Dancer (French-German Res. Inst. of Saint-Louis, 5 rue du général Cassagnou, BP 34, 68301 Saint-Louis Cedex, France, dancer@nucleus.fr)

This paper describes the electroacoustic components of a new active noise reduction (ANR) earplug device. Compared to a helmet, an earplug offers a better passive attenuation at low frequencies. Moreover, reducing dimensions of the acoustic cavity minimizes problems of resonance and acoustic delays, and allows active attenuation at higher frequencies (up to about 2 kHz). However, the microphone and the loudspeaker which are inserted into the earplug yield specific requirements (dimensions, output pressure, . . .). The microphone used is a commercially available Electret. The loudspeaker is specially designed for this application. It consists of a cylindrical piezoelectric ceramic loaded with a thin fluid air. This laterally radiating loudspeaker provides a high-volume flow compared to its dimensions. Prototypes with ceramics of different properties have been tested. A classic model of equivalent circuit based on electromechanical analogies has been adapted to describe their behavior. Theoretical and experimental frequency responses of the earplug in cavities, artificial and real ears will be presented. Lastly, several examples of simulation of the active attenuation that can be obtained with the earplug together with a numerical ANR filter designed by ISL will be mentioned. ^{a)}Also at Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen 72085 Le Mans Cedex 9, France.

3:20

5pSab5. Robust design of a sensorless bass-enhanced moving-coil ANC actuator. Mingsian R. Bai and Kwangmor Chu (Dept. of Mech. Eng., Natl. Chiao-Tung Univ., 1001 Ta-Hsueh Rd., Hsin-Chu 30050, Taiwan, ROC, msbai@cc.nctu.edu.tw)

Loudspeakers have been employed by far as the major ANC actuators. However, the responses of moving-coil loudspeakers are generally poor in low-frequency range because the diaphragms are unable to produce sufficient volume velocity. To alleviate the problem, this study focuses on

enhancing the low-frequency performance of loudspeakers by means of modern control techniques. Both magnitude and phase of the frequency response are optimally compensated. A self-sensing velocity observer is utilized for producing the cone velocity signal required by the controller. Feedback H_∞ robust control and feedforward H_2 model matching control are employed to simultaneously achieve robust stabilization and tracking performance. The proposed controller is implemented using a combined digital signal processor and operational amplifier circuitry.

3:40

5pSab6. Magnetostrictive actuation and digital variable structure control for active noise and vibration cancellation. R. D. Greenough, R. J. Smith (Dept. of Phys., Univ. of Hull, Hull HU6 7RX, UK), A. J. Wilkinson, and A. G. Jenner (Univ. of Hull, Hull HU6 7RX, UK)

Magnetostrictive actuators are ideally suited to active suppression of noise and vibration because of their low-frequency response, low-voltage operation, and high-load-bearing capability. In developing a system for active control, it was recognized that a strategy was important to contend with the nonlinear, hysteretic, and load-dependent behavior of the actuators. Consequently, digital variable structure control (DVSC) was selected, a method which has been successfully implemented in the laboratory and is now being employed and developed for practical applications. It has been demonstrated that with DVSC and a controller sample rate of 4 kHz, 32 dB of cancellation can be achieved at ~ 5 Hz. However, recent developments have increased the sample rate to ~ 8 kHz. Operation of the controller hardware with a simulated load has shown it capable of cancelling a vibration level of $0.001g_2/\text{Hz}_{0.5}$ up to 150 Hz. Experiments with the uprated controller have achieved cancellation levels of 40 dB in practice. In servo mode, the controller clearly demonstrates its ability to contend with the nonlinear properties of magnetostrictive actuators working under load. The system is presently being investigated for use in, for example, aircraft structures and industrial machines.

4:00

5pSab7. Piezo actuators for active vibration control and isolation—principles and practical experiences. Rolf Schirmacher (Müller-BBM GmbH, Robert-Koch-Straße 11, D-82152 Planegg, Germany, schi@mbbm.de)

Piezoelectric actuators based on PZT ceramics are a widely used drive system and are also often used for active noise and vibration control applications. They offer high static and dynamic forces, are very stiff, allow high-frequency dynamic applications, and contain no moving parts. But they also have severe disadvantages: They are very sensitive to bending moments and pulling forces and have to be driven by electrical fields of some kV/mm to obtain displacements of about 1%. The actuators have a high capacitance, which is a challenge for the power amplifiers, and the material exhibits hysteresis which shows up as nonlinearities. Based on these restrictions, different concepts for the application of piezo actuators for active vibration control and isolation systems are presented, and mechanical as well as electrical aspects are discussed. For a real life application, it was decided to use one of the concepts at an experimental stage. The design of this actuator system will be presented. Experimental results and experiences will be shown, including test stand results of the entire active vibration control system. The main problem proved to be with the nonlinearities of the actuator response, which are prohibitive for at least some applications.

Contributed Poster

This poster will be on display in the Poster Gallery from Thursday to Friday, 18–19 March. Author will be at the poster from 10:00 a.m. to 12:00 noon on Friday, 19 March.

5pSAb8. Test of various sound transducer systems for the use in ANC-systems and several applications of the ANC-system NANCY®. Dr. St. Deus and W. Schatz (ABS GmbH, Wildenbruchstraße 15, 07745 Jena, Germany)

With the introduction of ANC technologies, sound transducers have been increasingly applied in industrial fields, as well. In most cases, sound generators are set up through loudspeaker systems known from audio applications. Thus they are constructed to satisfy requirements of volume and quality (frequency response, distortion factor) while playing speech and music. This is different than applications of ANC systems in industrial

environments. The following features are mainly required: (1) production of high sound pressure, especially for low frequencies (down to 30 Hz and less); (2) resistance against high temperatures (up to 700 °C) and aggressive environments (high environment pressure or vacuum); (3) compact design; (4) long operating life (min. 1 year); and (5) directional effect (preferably unidirectional, sound downward). Currently, criteria as above are increasingly determining further developments of sound transducer systems. With respect to that, various sound transducer systems were tested in a lab channel with the ANC system NANCY®. The results are stated here. In addition several examples of use of the ANC system NANCY® are presented.

FRIDAY AFTERNOON, 19 MARCH 1999

POSTER GALLERY, 2:00 TO 6:00 P.M.

Session 5pSCa

Speech Communication: Speech Perception and Recognition (Poster Session)

James M. Hillenbrand, Chair

Speech Pathology and Audiology, Western Michigan University, Kalamazoo, Michigan 49008, USA

Contributed Papers

Posters will be on display in the Poster Gallery from Thursday to Friday, 18–19 March. Authors will be at their posters on Friday, 19 March. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 2:00 p.m. to 4:00 p.m. and contributors of even-numbered papers will be at their posters from 4:00 p.m. to 6:00 p.m.

5pSCa1. Vowel recognition from harmonic spectra. James M. Hillenbrand (Speech Pathol. and Audiol., Western Michigan Univ., Kalamazoo, MI 49008), Robert A. Houde (RIT Res. Corp., Rochester, NY 14623), Terrance M. Nearey (Univ. of Alberta, Edmonton, AB 26G 2E7, Canada), and Michael J. Clark (Western Michigan Univ., Kalamazoo, MI 49008)

Vowel classification methods based on either formant representations or overall spectral shape nearly always begin with the calculation of a smoothed, pitch-independent spectrum. In this report, a method is described for the classification of vowels directly from high-resolution spectra that retain harmonics of the voice source. Recordings were made of ten sustained vowels spoken by 20 men and 20 women. Smoothed spectral-shape templates for each of the vowels from a random half of the talkers were constructed as the average of the normalized spectra of like vowels spoken by different talkers. The key steps in the normalization procedure consist of: (1) removal of spectral tilt by the subtraction of a running average of spectral amplitudes, (2) dynamic range compression of the tilt-adjusted spectrum, and (3) smoothing across frequency. High-

resolution harmonic spectra from the remaining half of the talkers were then classified by computing the bin-for-bin difference between the harmonic spectrum and each of the smoothed spectral-shape templates.

5pSCa2. To be nasalized, or not to be nasalized. Ho-hsien Pan (Dept. of Foreign Lang. and Lit., Natl. Chiao Tung Univ., Hsinchu, Taiwan, 30005, ROC)

Vowel nasalization is a common process found in languages with or without phonemic contrasts between oral and nasal vowels. In English, there are no phonemic distinctions between oral and nasal vowels. Oral vowels are nasalized when followed by nasals. In Bengali and Hindi, where there are phonemic distinctions between oral and nasal vowels, oral vowels are nasalized when followed by nasals [Ferguson and Chowdhury (1960)]. In Taiwanese, which is a language with phonemic nasal and oral vowels, the presence and absence of nasality in the vowel nucleus is the major cue to lexical identification of words such as /saŋ-|bi-/ [saŋ-i] “send rice,” /saŋ-mi-/ [saŋ-mi-i] “send things” [Pan (1997)]. Thus the nasalization of Taiwanese oral vowels, which is a process based on the

“ease of articulation” principle, would impose difficulties for lexical identification. This study used both nasal airflow and acoustical data to investigate whether Taiwanese oral vowels are nasalized when both preceded and followed by nasals in words such as /tsin-¹gəŋ-/ [tsin-¹ŋəŋ-¹] “very silly.” The results showed that Taiwanese vowels are not nasalized when positioned between nasals. It is proposed that lexical identification interacts with phonetic process and prevents the vowel nasalization from occurring. [Work supported by NSC, Taiwan.]

5pSCa3. Voiced–unvoiced classification for recognition of stop consonants. Mahesh Krishnamoorthy, DongSuk Yuk, Krishna Dayanidhi, Samir Chennoukh, Daniel Sinder, and James Flanagan (CAIP Ctr., Rutgers Univ., 96 Frelinghuysen Rd., Piscataway, NJ 08854)

Significant error in stop consonant recognition is caused by the confusion between voiced stop consonants and their unvoiced counterparts. The recognition is based on HMM’s which use 12 MFCC’s and energy with their time derivatives. The voicing state is the distinctive feature of homorganic stop consonants. According to recognition error-rate analysis, it seems that the cepstral feature does not accurately represent the voicing state of the modeled phone. For this purpose, a voiced–unvoiced classifier in conjunction with HMM’s is proposed to improve the recognition of stop consonants. The recognition is done in two passes. In the first pass, a phone recognizer uses well-trained HMM’s to identify a stop consonant. This pass provides the recognized stop consonant in addition to the log probability. In the second pass, the voiced–unvoiced classifier checks if the voicing state of the phone segment matches with its phonetic description. In the case of mismatch and low probability of recognition, the voiced (unvoiced) consonant is swapped with its corresponding unvoiced (voiced) counterpart. Recognition results are presented in terms of error rate, using different techniques of voiced–unvoiced classification. This method reduces the recognition error rate of stop consonants. [Research supported by DARPA Contract Nos. DABT63-93-C-0037, N66001-96-C-8510, and NSF Contract No. IRI-9618854.]

5pSCa4. Speech segmentation and the use of sequential phonological constraints. Crouzet Olivier and Bacri Nicole (Laboratoire de Psychologie Experimentale, Universite Paris 5, CNRS URA 316, 28 rue Serpente, 75006 Paris, France, crouzet@psycho.univ-paris5.fr)

Previous experiments [Crouzet (1997); Vroomen and De Gelder (forthcoming)] led to the claim that constraints on legal and illegal consonant clusters may help to segment the speech signal. Two paradigms (word spotting and phoneme monitoring in nonwords) are contrasted in order to get a better understanding of (1) whether the previous results were actually linked to early phonological processes, to lexical competitions, or to distributional constraints and (2) whether the legality status of consonant clusters should be understood as the application of an abstract phonological knowledge on an already deciphered phonemic string or as a disruption of a speech rhythm that would increase the perceptual prominence of sound sequences and that could consequently take place before phonemic coding has been accomplished. The results of the word spotting experiment allows the argument that the effect should be interpreted in terms of phonological constraints. However, in the phoneme monitoring experiment, the reversed effect was observed. This fact could lead to the rejection of the speech rhythm disruption hypothesis. The phoneme monitoring task could not be relevant yet to investigate this issue and the use of another paradigm (speech perception in noise) that would be more appropriate is proposed.

5pSCa5. Burst spectra and place of articulation in read speech. Richard S. McGowan (Sensimetrics Corp., 48 Grove St., Somerville, MA 02144)

The burst spectra of voiceless stop releases are examined with a set of wide-band filters. These spectra are selected from one male speaker and one female speaker of American English, each reading from a list of 100 sentences. A set of rules is proposed for inferring place-of-articulation based on filter amplitudes obtained from a training set of utterances. The formulation of these rules is aided by segregating the data according to speaker and according to the second formant frequency at voice onset of the following vowel. This work is part of a project to infer articulatory movement from speech acoustics. [Work supported by the NIH through Grant No. DC-01247.]

5pSCa6. Does “one hundred fifty-five” mean 155 or 100,55 or 100,50,5? Corine Bickley (Res. Lab. of Electron., MIT, Rm. 36-579, Cambridge, MA 02139) and Lennart Nord (KTH, Stockholm, Sweden)

The goal of this study is to determine what are the acoustic cues, if any, that differentiate the spoken phrase that means 155 from one that means 100,55 or others for 150,5 or 1,155 or 100,50,5. The motivation for this study comes from an issue that arose in designing a speech-user interface—that is, a user interface controlled by a user’s speech via a speech recognizer. Users of a speech-user interface for programs that require numerical input (engineering design systems, accounting packages, etc.) need to speak sequences of numbers such as 100,50,5 (for example, to specify a three-dimensional coordinate location) as well as 100,55 (for a two-dimensional coordinate location). In this study, speakers produced utterances of one-, two-, and three-number sequences in two languages: Swedish and English. These productions were analyzed acoustically in terms of fundamental frequency, syllable duration, and amplitude. Listeners identified for each production whether the speaker intended to specify one, two, or three numbers, and which numbers. The results of the acoustic analyses will be presented and discussed with respect to the listener judgments. [Work supported in part by EC-TIDE Grant No. ENABL DE 3206.]

5pSCa7. The acoustic front-end of the FUL speech recognition system. Henning Reetz and Aditi Lahiri (Dept. of Linguist., Univ. of Konstanz, 78462 Konstanz, Germany, henning.reetz@uni-konstanz.de)

The featurally underspecified lexicon (FUL) speech recognition system converts the incoming speech signal into hierarchically organized monovalent phonological features in an early processing stage without identifying segments. After this conversion, only the presence of these features is used to access items from a 100 000 word dictionary which contains featural, segmental, morphological, and semantic information. The extraction of these features from the acoustic signal is based on standard LPC analysis for vowels, semi-vowels, and liquids, and from spectral shapes of DFT spectra for other sounds. Acoustic criteria for these features are only broadly specified to allow a wide range of speaker and dialectal variation. Most features are generated online from the signal without any further delay other than the width of the spectral windows. Only the feature [abrupt] uses some contextual temporal information.

5pSCa8. Inventory of the used “phonetic elements.” Shahla Sehhati (FSP-PV, Res. Ctr. for Network Technologies and Multi Media Applications Board, TUBKOM Tech. Univ. Berlin, TU Berlin, Sekr. MA 073 Straße des 17. Juni 136, D-10623 Berlin, Germany, shahla@prz.tu-berlin.de)

The extracted library of the phonetic elements for the speech synthesis based on phoneme cluster is extended additionally with the particularly selected elements. The auditive controlling observation and the experience by cluster segmentation have shown that the speech quality is improved by some critical phonetic combinations with the possibility of integration of

the important allophone variants in the library. This paper describes the reduction of a great number of these elements into an available limited quantity and the investigation of the effectivity of extending the library with selected allophones. Generally, the criteria for the selection of combined or hybrid basic elements for speech synthesis are one of the most interesting and challenging workfields. The subjective test results confirm that, in many cases, the addition of selected allophones optimizes further the quality of the library. With earlier results [Sh. Sehhati, Erstellen von Lautelementbibliotheken unter Verwendung von Phonemclustern auf der Grundlage des LPC-Sprachsyntheseverfahrens, dissertation, TU Berlin (1975)]. References: [L. Mackensen, *Deutsches Wörterbuch* (Südwest Verlag, München, 1967); Th. Siebs, *Deutsche Aussprache, Reine und gemäßigte Hochlautung mit Aussprache Wörterbuch* (de Gruyter, Berlin, 1969)].

5pSCa9. Production and perception of acoustic cues to gender and individual talker identity. Jo-Anne Bachorowski (Dept. of Psych., Wilson Hall, Vanderbilt Univ., Nashville, TN 37240, j.a.bachorowski@vanderbilt.edu) and Michael Owren (Cornell Univ., Ithaca, NY 14853)

Although listeners routinely identify both gender and individual identity of talkers from their speech, explanations of these abilities remain problematic. Integrating source-filter theory with knowledge regarding variation in vocal production-related anatomy, predictions concerning gender and talker indexical cueing were tested using "point," /e/, and schwa vowels that were extracted from the naturally produced speech of 40 male and 40 female talkers. Very high rates of gender classification were obtained using F_0 and an estimate of vocal tract length, the two measures that best reflect sexual dimorphism in vocal-production-related anatomy. In contrast, individual talker classification depends primarily on the various cues associated with vocal tract filtering. Ongoing perceptual studies are being conducted to evaluate the perceptual importance of these findings to listeners. [Work supported by NSF.]

5pSCa10. The time course of lexical involvement in phonetic categorization. James M. McQueen (Max-Planck-Inst. for Psycholinguist., Wundtlaan 1, 6525 XD Nijmegen, The Netherlands), Dennis Norris (MRC Cognition and Brain Sci. Unit, Cambridge CB2 2EF, UK), and Anne Cutler (Max-Planck-Inst. for Psycholinguist., 6525 XD Nijmegen, The Netherlands)

Two experiments in Dutch examined how the influence of lexical knowledge on phonetic decisions changes over time. A [f]–[s] continuum (edited natural speech) was placed in initial position in monosyllables to make word–nonword and nonword–word continua: *flauw* (dull)–*slauw* and *flaap*–*slaap* (sleep). Syllable-final continua were made in the same way: *maf* (silly)–*mas* and *jaf*–*jas* (coat). Materials were low-pass filtered at 3 kHz. In experiment 1 there was a significant lexical effect for both initial and final fricatives: throughout the continua there were more [f] responses in the word–nonword than in the nonword–word continua. Lexical involvement was strongest in the listeners' fastest responses but disappeared in their slowest responses, both for the final fricatives (in line with previous findings) and the initial fricatives (contradicting earlier studies showing the strongest effects in the slowest responses). Experiment 2 tested whether listeners would still use lexical knowledge in their fastest

responses under severe time pressure. They were asked to respond before a tone, presented 500 ms after fricative offset. The results replicated experiment 1. Lexical knowledge appears to be used in phonetic decision-making only within a limited time frame.

5pSCa11. Use of temporal envelope cues for the recognition of words and phonologically significant contrasts. Liat Kishon-Rabin and Michal Nir-Dankner (Commun. Disord. Dept., Sackler School of Medicine, Tel-Aviv Univ., Ramat-Aviv, Israel, lrabin@ccsg.tau.ac.il)

The goals of this study were: (1) measure the recognition of words and speech pattern contrasts using temporal cues only, and (2) to evaluate the effect of training on the ability to perceive segmental, suprasegmental, and phoneme information from the speech waveform only. Fifteen subjects, 20–40 years of age, with normal hearing, were tested before and after extensive training. Test stimuli consisted of: (a) eight segmental and two suprasegmental contrasts of the SPAC test using a binary forced-choice paradigm, and (b) monosyllabic words from the isophonemic AB word lists presented in open set. Spectral information was eliminated by multiplying the speech waveform with white noise using commercially available array processing software. Testing and training were under computer control. Preliminary results suggest that: (1) intonation and stress are perceived well using temporal information only, (2) manner is perceived better than either place or voicing contrasts, (3) training improves the perception of speech using temporal cues only, and (4) there is a strong correlation between speech pattern contrasts as measured in open and closed set paradigms. The findings will be discussed in relation to the relative importance of the stimuli's spectral and temporal cues and their relevance to clinical data of individuals with profound hearing loss.

5pSCa12. On the link between acoustic cue distributions and categorization dependencies. Roel Smits (Max Planck Inst. for Psycholinguist., Wundtlaan 1, 6525 XD Nijmegen, The Netherlands, Roel.Smits@mpi.nl)

The HICAT model of hierarchical categorization [R. Smits, J. Acoust. Soc. Am. **103**, 2980(A) (1998)] explicitly models dependencies in four-alternative forced-choice categorizations involving two binary distinctions. A philosophy has been developed which predicts which dependencies are likely to occur for a given pair of phonetic distinctions on the basis of the acoustic distributions of relevant cues in natural utterances. It is argued that, because speech perception happens under severe time pressure, listeners try to minimize categorization complexity while maximizing categorization accuracy. For certain phonetic distinctions, the cue distributions allow listeners to use simple, independent categorization strategies while still performing close to optimally. Severe coarticulation, however, necessitates a more complex strategy involving categorization dependencies which reflect dependencies in the cue distributions. This philosophy was tested experimentally. Spectral locations of fricative and vowel resonances in syllables /si, sy, Si, Sy/ were measured on a large set of naturally spoken tokens. Based on the resulting distributions it was predicted that listeners' fricative categorization will be dependent on the vowel categorization. Next, a categorization experiment was run using a two-dimensional fricative-vowel continuum. The dependencies inferred from the categorization data using the HICAT model will be compared to the predictions.

Session 5pSCb

Speech Communication: Consonants and Vowels (Poster Session)

Carol Espy-Wilson, Chair

ECE Department, Boston University, 8 St. Mary's Street, Boston, Massachusetts 02215, USA

Contributed Papers

All posters will be on display in the Poster Gallery from Thursday to Friday, 18–19 March. Authors will be at their posters on Friday, 19 March. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 2:00 p.m. to 4:00 p.m. and contributors of even-numbered papers will be at their posters from 4:00 p.m. to 6:00 p.m.

5pSCb1. Variation in the pronunciation of German “ein_.” Peter M. Janker, Bernd Pompino-Marschall, and Seadet Zeynalowa (Ctr. for General Linguist., Berlin, Germany)

Pronunciation variation of German “ein_” is examined (unstressed indefinite article “ein_,” “einen,” as well as stressed instances of “ein_,” i.e., the numeral or as part of a compound) based on a larger corpus of read and spontaneous speech according to segmental variation (elisions, assimilations, etc.) and acoustic features (segmental durations, extent of formant transitions, etc.). Second, the results of a combined electropalatographical/electromagnetic-articulographical study are presented. Two male subjects uttered the items “ein_” and “einen” embedded in test sentences at normal, slow, and fast speech rate. Electropalatographically tongue palate contacts for the nasal segments (duration, center of gravity of contact area) were analyzed. The EMA measurements of movement from the diphthong to nasal (or from diphthong to nasal to schwa to nasal) were done with a sensor coil mounted 1 cm behind the tip of the tongue and another one mounted on a strip of elastic foil glued to the EPG palate for sensoring velar movement. Articulatory analysis revealed quite complex gestural reductions for bisyllabic “einen.” Even for items without schwa elision (only at a slow rate) there is no velar closing movement for the schwa.

5pSCb2. Taiwanese prenasalized stops and poststopped nasals. Hsien Pan (Dept. of Foreign Lang. and Lit., Natl. Chiao Tung Univ., Hsinchu, Taiwan, 30050, ROC)

Prenasalized stops and poststopped nasals were classified into a category of “nasal+stop” segments in Chinese [Chan (1987)]. In Taiwanese, which is a member of the Chinese language family, initial prenasalized stops b.g [m⁰b, ⁹g] would change into homorganic nasals [m, ⁹g] when preceded by nasals. Moreover, strong energy attacks can be observed at the release of oral closures for b.g [m, ⁹g] [Pan(1994)]. This raises the issue of whether Taiwanese prenasalized stops alternate with post stopped nasals. This study investigated the alternation between prenasalized stops and post stopped nasals in Taiwanese by using nasal airflow, oral airflow, and acoustical data. Preliminary results showed that when prenasalized stops changed into nasals, there was energy damping before the release of oral closure, and rise of acoustical energy at the release of oral closure. Though not environmentally conditioned, post stopped nasals [m⁰b, ⁹g] were free variations of homorganic prenasalized stops [m^p, ⁹g^k]. [Work supported by NSC, Taiwan.]

5pSCb3. Do harmonic consonant clusters in Georgian have only one release? Priscilla C. McCoy (Phonology Lab., Univ. of California, Berkeley, CA 94720)

It has been claimed that in Georgian, a language in the Kartvelian or South Caucasian family of languages, harmonic consonant clusters have a single release. To examine this claim, different constituencies and orderings of harmonic consonant clusters and nonharmonic consonant clusters were recorded and analyzed in order to (1) assess the phonetic reality of the initial claim; and (2) compare durations of cluster releases and whole cluster segments in nonharmonic environments to those, if any, exhibited in harmonic environments. Preliminary results of the relative distribution of release durations across these two groups illustrate an interesting division of release and cluster duration figures and (2) show a possible positive correlation of a sonority hierarchy [G. N. Clements, “The role of the sonority cycle in core syllabification,” in *Between the Grammar and Physics of Speech*, edited by J. Kingston and M. E. Beckman, Papers in Laboratory Phonology 1 (Cambridge U.P., Cambridge, 1990)]. This also builds on previous work on Georgian consonants and their relationship to a sonority hierarchy by McCoy [1995]. The implications for consonant clusters and Georgian phonology are investigated, which in turn suggests some phonological priorities in Georgian.

5pSCb4. Effect of frequency modulation of F0 on the accuracy of formant frequency matches. Pascal Dissard and Chris J. Darwin (Lab. of Exp. Psych., Univ. of Sussex, Brighton, Sussex BN1 9QG, UK)

Listeners adjusted the formant frequency of a one-formant complex sound to match the timbre of a similar fixed sound (formant frequency = 1100 or 1200 Hz). On each trial the two sounds' fundamental frequencies (F0) were in the ratio 1 : 1.13. In half the blocks the sounds were on low fundamentals (80 and 90.4 Hz), giving unresolved harmonics in the region of the formant peak; in the other half of the blocks the F0 was high (221.2 and 250 Hz), giving resolved harmonics near the formant peak. Orthogonal to this manipulation the F0 was either unmodulated for both sounds, or had a frequency modulation (FM) of 6 Hz and 3% depth. Data from seven listeners showed that, with no FM, matches are more variable within each listener for sounds with resolved than for sounds with unresolved harmonics, but that this difference disappears with FM. The result provides evidence that frequency modulating F0 improves listeners' ability to estimate a formant frequency that is cued by resolved harmonics. [Work supported by EPSRC Grant No. GR/L03422.]

5pSCb5. Articulatory discriminability of vowels: Articulator and corpus effects. Philip Hoole (Inst. fuer Phonetik, Munich Univ., Schellingstr. 3, D-80799 Munich, Germany, hoole@phonetik.uni-muenchen.de)

The principal aim of this work was to compare the potency of different sources of articulatory information for differentiating the vowels of German in discriminant analyses. The secondary aim was to determine whether results depended on the corpus analyzed. Articulatory data (electromagnetic articulography; seven speakers) was available for tongue, lower-lip, and jaw position. Three corpora were compared. The first two used highly controlled nonsense-word material but differed in speech rate (normal versus fast). The third corpus embedded the vowels in a wide variety of real words and sentences. Using as a baseline the classification accuracy obtained with tongue-position data only, the increase in accuracy was about 9% with jaw data, 14% with lip data, and 17% with both jaw and lip. A further noticeable improvement (23% re. baseline) resulted from adding in velocity or acceleration data. Classification patterns were remarkably similar over the three corpora; the real-word corpus was simply, but consistently, about 10% worse than the “nonsense-normal” corpus, with “nonsense-fast” in between. Results will be related to the structure of the German vowel system. Corpus effects on articulatory behavior will be further discussed by comparing PARAFAC factor analysis of tongue configuration for the nonsense versus the real corpora. [Work supported by DFG Ti69/29.]

5pSCb6. Perceptual consequences of unintended epenthetic stops. Natasha Warner and Andrea Weber (Max Planck Inst. for Psycholinguist., Nijmegen, The Netherlands)

In many languages, historically an epenthetic stop has appeared between a nasal and a fricative or a nasal and another stop, as /p/ in English “empty” (from earlier “æmtig”). Ohala [Phonetica 52, 160–170 (1995), and in press] proposes an articulatory explanation for epenthesis involving listener misperceptions. Ohala predicts that epenthetic stops will match the place of articulation of the preceding nasal. The current study investigates listeners’ perception of epenthetic stops. A speaker produced nonsense words ending in nasal stop sequences, such as /flg@mk, tuloʔp/ from orthographic representation. Thus the speaker did not intend to produce epenthetic stops [(floʃmpk, tuloʔkp)]. Listeners monitored for /p/ or /k/. Listeners perceived a /p/ significantly more often after labial nasals (more /p/ responses to /mk/ final words than to /nk/ and /rhoonk/ final words, $p < 0.0001$), and perceived /k/ significantly more often after velar nasals (more /k/ responses to /loʔp/ final words than to /mp/ and /np/ final words, $p < 0.0001$). Tokens with epenthetic stop bursts (acoustic analysis) received the most responses. These results provide experimental verification of the proposed articulatory and perceptual origins of epenthetic stops: listeners perceive stops the speaker did not intend, predominantly in environments where the articulatory explanation predicts epenthetic stops to occur.

5pSCb7. The relevance of F_4 in distinguishing between different articulatory configurations of American English /r/. Carol Y. Espy-Wilson and Suzanne Boyce (Boston Univ., ECE Dept., 8 St. Mary’s St., Boston, MA 02215)

American English /r/ is often cited as a segment exemplifying a many-to-one articulatory to acoustic relationship. Articulatory data show a continuum of vocal tract configurations for /r/, ranging from “retroflexed” to “bunched,” all involving three constrictions: (1) in the pharynx, (2) along the palate, and (3) at the lips. Researchers have noted, however, that all of these configurations result in similar acoustic profiles, at least for F_1 , F_2 , and F_3 . In this study, acoustic and articulatory data were used to investigate the mechanisms which determine the shape of the F_4 trajectory for /r/. Articulatory positioning data and simultaneous acoustic data were collected from five subjects who said the phrase “Say warav for me.” The Electromagnetic Midsagittal Articulometer (EMMA) system was used to record the positions of three points on the tongue. Preliminary analysis

based on the y position of the front, mid, and back transducers suggests that the behavior of F_4 changes with different /r/ configurations and that the frequency of F_4 is based on the length of the cavity between the palatal and pharyngeal constrictions.

5pSCb8. Vowel-dependent VOT variation: An experimental study. Steve S. Chang, John J. Ohala, Gunnar Hansson, Benjamin James (UC Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94720, changs@socrates.berkeley.edu), Julie Lewis, Lily Liaw, Margaret Urban, Alan Yu (UC Berkeley), and Kenneth Van Bik (UC Berkeley)

That the VOT of voiceless stops is typically longer before high, close vowels and shorter before low, open ones is well documented. The hypothesis was tested that high vowels engender longer VOT because they offer greater resistance to the air escaping from the mouth (thereby delaying the transglottal pressure differential required for voicing). A correlation was sought between oral pressure decay (OPD) and VOT. Acoustic and aerodynamic data were collected from a male native speaker of American English uttering 12 nonsense words of the type atɪ, atə in isolation ten times. Pressure data was sampled by inserting a tube into the subject’s pharyngeal cavity via the nose. The VOT was measured from the release of the burst to the onset of the first periodic cycle. The OPD was defined as the time required for oral pressure to decrease from the peak to a fixed low threshold. The OPD and VOT measurements displayed a significant, linear correlation, indicating that VOT and vowel height are indeed mechanically linked. Moreover, tokens with voiceless stops preceding sonorants, e.g., atwà, akjà, yielded the longest VOTs since sonorants have narrower constrictions, resulting in longer OPDs.

5pSCb9. The perception of German syllabic [n]. Bernd Pompino-Marschall and Peter M. Janker (Ctr. for General Linguist., Berlin, Germany)

Perception of syllabic [n] was studied using different inflectional forms of the German indefinite article, i.e., “ein” (nom. sing. masc.) versus “einen” (acc. sing. masc.). Inspection of a larger corpus of spoken German revealed that the most frequent pronunciation of “einen” is the allegro form with only a lengthened—not literally bisyllabic—[n]. Identification tests were run with manipulated items of a naturally spoken “einen.” The original utterance consisted of a glottalized [a] segment of 68-, a diphthong of 65-, and a [n] of 49-ms duration produced at a fundamental frequency of about 200 Hz by a female speaker. The duration of the diphthong and the nasal were varied in five steps of approximately 10 ms by doubling/cutting pairs of individual pitch periods. A second set of 25 stimuli was produced by cutting the initial glottalized segment. Analyses of variance revealed highly significant effects of nasal duration and glottalization independent of diphthong duration: Longer nasal segments as well as the presence of glottalization resulted in an increase in “einen” responses. The identification seems to be independent of rate of articulation but not of general speaking rate when the test items are embedded in larger utterances.

5pSCb10. Loudness and prominence in front vowels. Frank G. Gooding (Dept. of Linguist., Univ. of Wales, Bangor LL57 2DG, UK)

The perception of the high front vowels such as [i] is known to be dominated by the upper formant (UF) region, as is shown by the fact that [i] can be elicited by a single formant positioned there [P. Delattre *et al.*, Word 8, 195–210 (1952)]. The basis of this dominance remains unclear, however, and is investigated by the present study. The simplest explanation would be that the UF region dominates F_1 in terms of peak or total loudness. However, this is not reflected in current auditory models [e.g., B. Moore and B. Glasberg, Acustica 82, 335–345 (1996)]. The representations of [i]-like vowels produced by this model, for instance, whether expressed as dB, phons, or sones, all show F_1 strongly dominating the UF region in terms of excitation level or loudness. Two possible explanations

for this discrepancy come to mind: first, that the model is inaccurate; or second, that the perceptual dominance reflects mechanisms more central to the modeled excitation patterns. In the first case, the inaccuracy is likely to lie in the equal loudness data used by the model. An experiment was designed to test between these possibilities. Two formant versions of [i] and other front vowels were presented to subjects, who could control the amplitude of F_2 . The F_2 threshold and equal loudness point was determined for each subject. Results indicate that the equal loudness point of F_2 occurs at an intensity far below, and hence reflects a sensitivity far greater, than that predicted by the model. This implies that central mechanisms are not required to explain the perceptual dominance, and that the equal loudness data probably needs to be revised.

5pSCb11. Adults' perception and production of the English vowel /i/. Elaina Frieda^{a)} (Bussenstr. 32, 70184 Stuttgart, Germany, emfrieda@netscape.net)

The present experiment examined the link between the perception and production of the English vowel /i/. Thirty-five male subjects produced the vowel /i/ in the following two conditions: first, subjects produced "casual" or citation speech; second, subjects produced exaggerated (hyperarticulated) speech. Subjects then completed a perceptual experiment employing a method of adjustment procedure which enabled them to select their own preferred or ideal exemplar for /i/. Subjects' vowel productions were measured with LPC analysis for the following parameters: F_0 , F_1 , F_2 , and duration. The perceptual data were averaged for each subject to determine their preferred F_1 and F_2 dimensions. Comparing the two speech samples yielded predicted results in that the hyperarticulated speech was more extreme and fronted within the vowel space than the citation speech. Analyses also revealed a negative correlation between subjects' citation speech and their perceptual data, reflecting a tendency for the perceptual data to be higher and more fronted in the vowel space than citation speech. The production data were converted to bark difference scores for $F_1 - F_0$ (vowel height) and $F_2 - F_1$ (front-back dimension). These values were compared to subjects' perceptual data and demonstrated that the hyperarticulated speech was more similar to the perceptual data than citation speech.^{a)} Formerly at Dept. of Psych., Univ. of Alabama, Birmingham, AL.

5pSCb12. Reflection patterns and coloration of vowels. Anthony J. Watkins (Dept. of Psych., Univ. of Reading, Reading RG6 6AL, UK, syswatkn@reading.ac.uk)

When sounds are played in a room, the direct sound is accompanied by later-arriving reflections. If the reflections arrive within about 50 ms they are not heard separately, but seem to be perceptually incorporated with the direct sound. In addition, the reflection pattern can have a filtering effect, which can change the sound's spectral envelope, and which might be heard as a change in timbre or "coloration." Thus, colored reflected sounds tend to follow after an uncolored direct sound. It has been suggested that the reflections' coloration is perceptually suppressed under these circumstances. The present experiment was designed to test this idea. Brief, 20-ms reflection patterns were designed that "colored" vowels' spectral envelopes and which followed after a direct sound in "two-part filters." Coloration was measured by changes in listeners' identifications of vowels that are played through these filters. This was compared with the coloration in a condition with reversed two-part filters where the reflection pattern was caused to precede the direct sound. There should be more coloration in the reversed condition if there is normally suppression of coloration from later-arriving reflections. However, coloration was no greater in the reversed condition even though the reflections' perceptual effects were substantial.

5pSCb13. Help or hindrance: How violation of different assimilation rules affects spoken-language processing. Andrea Weber (Max-Planck-Inst. for Psycholinguist., P.O. Box 310, 6500 AH Nijmegen, The Netherlands, andrea.weber@mpi.nl)

Recent phoneme detection studies showed that spoken-language processing is inhibited by violation of obligatory assimilation [e.g., Otake *et al.* (1996)]. A phoneme detection experiment was designed to replicate this effect with German fricative assimilation. German fricative assimilation is progressive and applies within syllables. The velar fricative [x] occurs after back vowels, the palatal fricative [ç] after front vowels. In contrast to previous findings, listeners detected [x] *faster* when violation occurred. A second experiment explored whether these results are due to the tested assimilation applying within syllables, whereas earlier experiments tested assimilation across syllables. Again, listeners detected [x] *faster* when the assimilation rule was violated across a syllable boundary. The discrepancy with the earlier results might be due to the German fricative assimilation rule being progressive, while earlier experiments tested regressive assimilation. A third experiment tested German place assimilation for nasals. Regressive place assimilation in German is obligatory within syllables. A velar stop /k/ specifies the place for the preceding nasal /ŋ/. This time listeners detected the target phoneme /k/ *more slowly* when the assimilation was violated. The results show that whereas violation of regressive assimilation inhibits processing, violation of progressive assimilation speeds up processing.

5pSCb14. Perceptual properties of English fricatives. Allard Jongman, Michelle Spence, Yue Wang, Brian Kim, and Diana Schenck (Cornell Phonet. Lab., Dept. of Modern Lang., Cornell Univ., Ithaca, NY 14853)

A series of experiments was conducted to determine the properties that contribute to fricative perception. Listeners' identification of English fricatives based on fricative-vowel syllables and on isolated fricative noise portions reveals the perceptual salience of each fricative and the extent to which fricative-to-vowel transitions contribute to identification. Two further experiments specifically address perception of the nonsibilant fricatives, which, it has been claimed, may be based more on semantic or facial factors. One experiment investigates how a semantically matching or mismatching precursor affects perception of minimal pairs (e.g., fin-thin). A second experiment determines the extent to which auditory and/or visual information each contribute to fricative perception. Moreover, regression of these perceptual data on acoustic measurements, including noise duration, amplitude, relative amplitude, spectral peak location, spectral moments, and locus equations [see Jongman *et al.*, J. Acoust. Soc. Am. **103**, 3086A (1998)], indicates which acoustic properties play a significant role in fricative perception. [Work supported by NIH.]

5pSCb15. Influence of frequency range in the perceptual recognition of fricatives. Sergio Feijóo, Santiago Fernández, and Ramón Balsa (Departamento de Física Aplicada, Universidad de Santiago de Compostela, 15706 Santiago, Spain, fasergio@usc.es)

The objective of this paper is to study the importance of various frequency bands for the identification of fricatives. Tokens were CV syllables formed by the combination of the Galician fricatives /θ, f, s, ʃ/ and the vowels /a, e, i, o, u/ which were pronounced in Hyperspeech form by a man and a woman. Tokens were sampled at 32 kHz and low-pass filtered with cutoff frequencies of 11, 8, 5.5, 4, and 3 kHz. Thus, the total number of tokens was $240 = 4 \text{ fricatives} \times 5 \text{ vowels} \times 2 \text{ sexes} \times 6 \text{ frequencies}$. Thirty-seven listeners carried out the perceptual experiments in two conditions: (1) whole fricative noise plus 100 ms of the following vowel, and (2) whole fricative noise. The results of the perceptual experiments show that as the cutoff frequency is lowered, (a) /s/ tends to be recognized as /θ/ in both conditions; (b) the fricative noise of /θ/ tends to be recognized as /f/, and (c) recognition of /f/ and /ʃ/ is affected to a lesser extent. Results

suggest the importance of low-frequency energy in the characterization of /f/ and /ʃ/. Acoustic analysis is being carried out to determine if the identification of the filtered fricatives can be explained by relations among the spectral bands.

5pSCb16. Phonetic priming of features in a naming task. Tobey L. Doeleman (Cornell Phonet. Lab., Dept. of Linguist., Cornell Univ., Ithaca, NY 14853, tld5@cornell.edu)

The role of features in speech perception is investigated in a series of priming experiments in which subjects named CV syllables consisting of the English consonants [p, b, t, d, f, v, s, z] followed by the vowel [a]. Natural tokens were used to create sixty-four prime-target pairs which varied critically the number of features the consonants shared (0–3). The pairs were further classified into 8 stimulus sets based on the feature relationship: unrelated, identity, shared manner and voicing, manner and place, voicing and place, manner only, voicing only, and place only. Patterns of reaction times and accuracy data for the naming task are compared for the different stimulus sets. The same 64 pairs were blocked according to the voicing, place, or manner feature of the target, and presented to three subsequent groups of subjects who were thus given information about a particular feature. Results are discussed with reference to previous findings from a series of gating experiments (using the same stimulus set manipulation) which found that subjects given place information had significantly higher scores in a consonant identification task than subjects given manner information [T. L. Doeleman, J. Acoust. Soc. Am. **101**, 3111(A) (1997)].

5pSCb17. Top-down influences on phonetic processing as a function of aging. Jeffrey Boyczuk and Shari Baum (McGill Univ., School of Commun. Sci. and Disord., 1266 Pine Ave. W., Montreal, QC H3G 1A8, Canada, jboyucz@po-box.mcgill.ca)

A phonetic identification experiment was conducted in order to assess the extent to which lexical information influenced low-level speech perception processes in three different age groups of subjects (20–29 years, 60–69 years, and 70–79 years). Subjects were required to label syllable-initial, bilabial stop consonants which varied in voice onset time (VOT) as either /b/ or /p/. Following Ganong [J.E.P.: H.P.P. **6**, 110–125 (1980)], a pair of VOT continua with opposing lexical biases was created: one continuum ranged from the word /buk/ to the nonword /puk/ while the other continuum ranged from the nonword /bu t/ to the word /put/. VOT values for the syllable-initial /b/–/p/ phoneme were the same at corresponding steps in the two continua. Results indicated a significant influence of lexical status on phonetic identifications for all age groups. Furthermore, a significant difference in the location of the phonetic crossover boundary was observed, with older subjects showing longer VOT values than young subjects. While findings suggest that top-down processing is still a preva-

lent part of phonetic perception in elderly individuals, it appears that aging may bring changes in the values of the acoustic parameters used to define phonetic categories. [Work supported by McGill University.]

5pSCb18. Speaking rate effects on vowel identification in natural sentence contexts. Adam M. Berman, Jonathan S. Neville, and Terry L. Gottfried (Dept. of Psych., Lawrence Univ., Appleton, WI 54912)

Previous research [T. L. Gottfried, J. L. Miller, and P. E. Payton, *Phonetica* **47**, 155–172 (1990)] has indicated for synthetic speech that vowel identification changes according to the speaking rate of the sentence context, even when the vowel contrast is specified experimentally only by spectral differences. This study tests listeners' identification of /i/ vs /ɪ/ in stimulus continua created by altering formant values and durations of the target syllables "beat" and "bit" in naturally produced sentences. These targets were spoken in two sentence frames, one spoken at a moderate rate of speech and the other compressed to twice the speaking rate. Longer syllable durations significantly shifted the boundary of the spectral continuum, so that there were more "beat" responses as compared to shorter durations. However, unlike what was found for synthetic vowel series, there was no significant effect of sentence rate on the number of "beat" responses. Reasons for this different outcome, and the relevance of these findings for theories of rate normalization, are discussed. [Work supported by Lawrence University Faculty Grant.]

5pSCb19. Perceptual vowel spaces derived by computer game. James D. Miller, Arnold F. Heidbreder, and Nicole N. Wyzinski (Central Inst. for the Deaf, 818 S. Euclid Ave., Saint Louis, MO 63110, jdm@cid.wustl.edu)

A same–different task embedded in a computer game is used to obtain latencies of responses to vowel pairs. The "player" responds as fast as possible when the sounds in a pair are different and is not to respond when they are the same. The game concept and computer program were originally developed by K. Manabe and R. Dooling of the University of Maryland. Vowel sounds /i,ae,a,u/ were synthesized with interformant valley depths of 9, 18, 36, and 99 dB. All vowels were equalized in loudness based on balances made by four normally hearing listeners. Five listeners, two with normal hearing, two hearing-aid users, and one cochlear-implant user were tested four times, once at each valley depth. The logarithmic mean latencies for each different pair were taken as similarity measures for multidimensional scaling (mds). Average latencies decreased with increasing valley depths. For the two normally hearing listeners and one of the hearing-aid users, the perceptual vowel spaces derived by mds looked like the vowel equilateral, and the intervowel distances correlated highly with distances in Millers Auditory Perceptual Space. For one hearing-aid and the cochlear-implant user the vowel spaces were abnormal in appearance. Individual results will be presented.